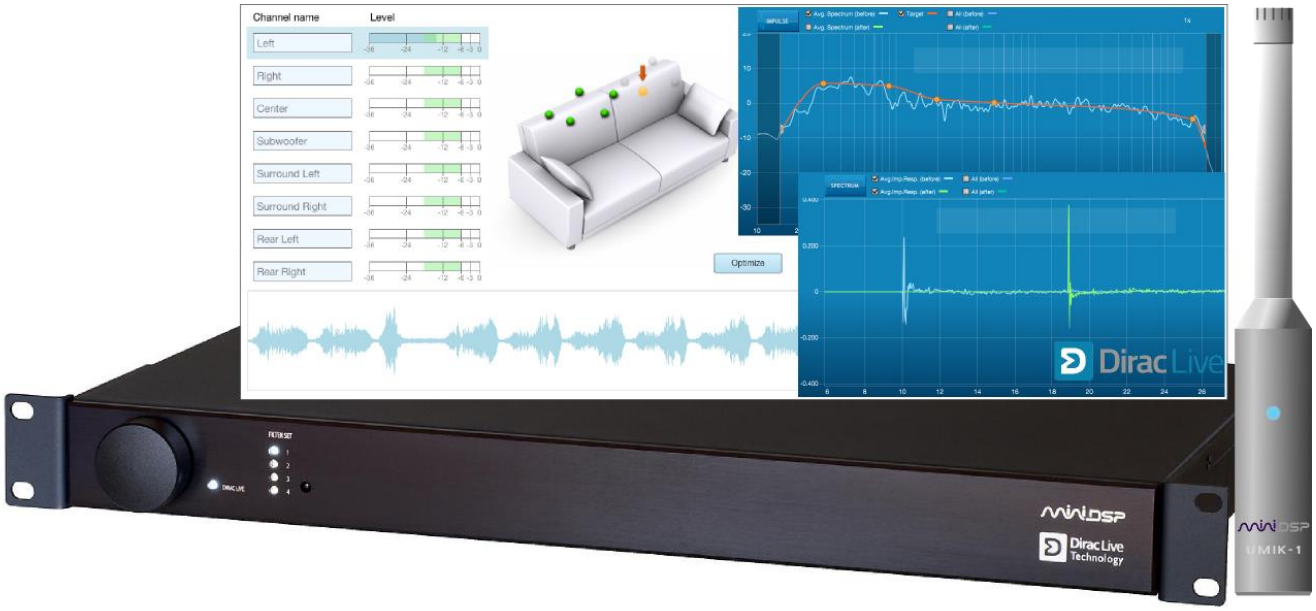
DDRC-88A

8-CHANNEL AUDIO PROCESSOR

WITH DIRAC LIVE® TECHNOLOGY

User Manual



Revision history

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0.2	Preliminary pre-release version	5 December 2014
0.3	Draft pre-release for comment	20 January 2015
0.4	Second draft pre-release for comment	28 January 2015
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1.1	Updated firmware update procedure	20 May 2015

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IMPORTANT INFORMATION

Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at minidsp.desk.com.

SYSTEM REQUIREMENTS

To configure your *DDRC-88A* audio processor, you will require a Windows PC with the following minimum specification:

- Intel Pentium III or later, AMD Athlon XP or later
- 2 Gigabytes (GB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- Microsoft® Windows® Vista® SP1/Win7/Win8
- Microsoft® .NET framework v3.5 or later
- Two free USB 2.0 ports

DISCLAIMER/WARNING

miniDSP cannot be held responsible for any damage that may result from the improper use or incorrect configuration of this product. Please read this manual carefully to ensure that you fully understand how to operate and use this product, as incorrect use or use beyond the parameters and ways recommended in this manual have the potential to cause damage to your audio system.

Please also note that many of the questions we receive at the technical support department are already answered in this User Manual and in the online application notes on the miniDSP.com website. So please take the time to carefully read this user manual and the online technical documentation. And if an issue arises with your unit, please read through the Troubleshooting section first. Thank you for your understanding!

WARRANTY TERMS

miniDSP Ltd warrants this product to be free from defects in materials and workmanship for a period of one year from the invoice date. Our warranty does not cover failure of the product due to incorrect connection or installation, improper or undocumented use, unauthorized servicing, modification or alteration of the unit in any way, or any usage outside of that recommended in this manual. If in doubt, contact miniDSP prior to use.

FCC CLASS B STATEMENT

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

- This device may not cause harmful interference.
- This device must accept any interference received, including interference that may cause undesired operation.

Warning: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Notice: Shielded interface cable must be used in order to comply with emission limits.

Notice: Changes or modification not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

CE Mark Statement

The DDRC-88A has passed the test performed according to European Standard EN 55022 Class B.

Package Contents

Your *DDRC-88A* package includes:

- One *DDRC-88A* audio processor
- One universal 12 VDC power supply
- One USB cable for computer connectivity (1.5m)
- One UMIK-1 calibrated measurement microphone with USB cable
- One tripod microphone stand
- One full license for **Dirac Live Calibration Tool for miniDSP**

A Note on this Manual

This User Manual is designed for reading in both print and on the computer. If printing the manual, please print double-sided. The embedded page size is 8½" x 11". Printing on A4 paper will result in a slightly reduced size.

For reading on the computer, we have included hyperlinked cross-references throughout the manual. In addition, a table of contents is embedded in the PDF file. Displaying this table of contents will make navigation much easier:

- In Adobe Reader on Windows, click on the "bookmarks" icon at the left. The table of contents will appear on the left and can be unfolded at each level by clicking on the "+" icons.
- In Preview on the Mac, click on the **View** menu and select **Table of Contents**. The table of contents will appear on the left and can be unfolded at each level by clicking on the triangle icons.

1 Product Overview

Thank you for purchasing a miniDSP *DDRC-88A* audio processor powered by Dirac Live®, the world's premier room correction solution. We are delighted to offer you this software and hardware combination, the fruit of extensive research and development and years of experience in sound system tuning.

The *DDRC-88A* is an 8-channel digital audio signal processor (DSP) running the Dirac Live® room correction algorithm. The onboard floating-point SHARC processor provides time and frequency correction of a 7.1 monitoring, home theater or multichannel audio system. Inputs and outputs are analog, available as both single-ended signals connected via RCA jacks, and as balanced signals via Phoenix terminal blocks.

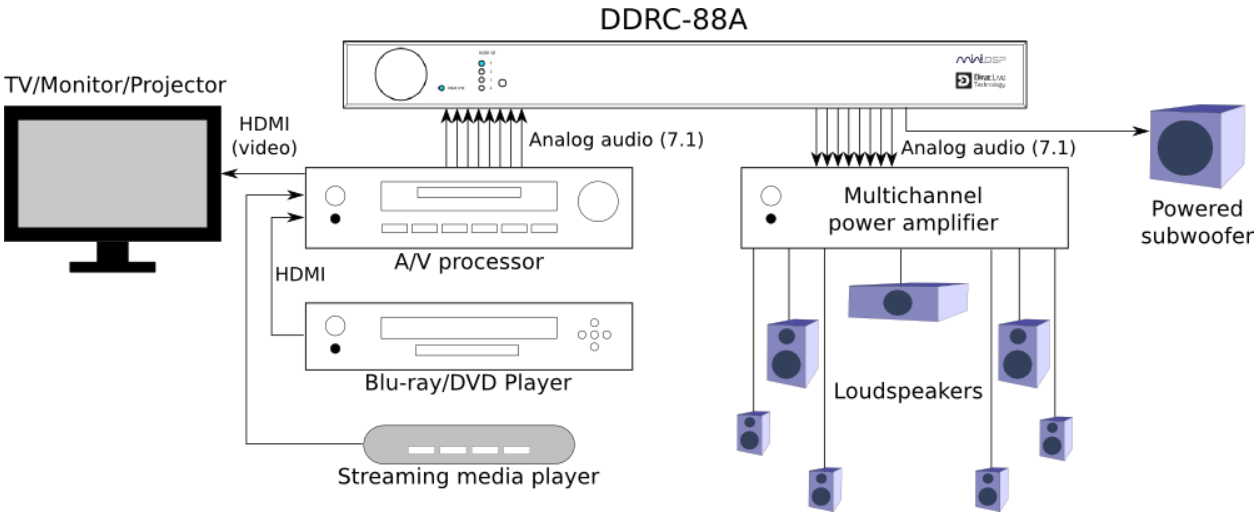
The *DDRC-88A* is a member of the miniDSP *Dirac Series* of audio processors. Deploying a *DDRC-88A* will:

- Improve imaging and immersion
- Improve clarity of music and dialog
- Produce a tighter bass
- Reduce listening fatigue
- Remove resonances and room modes

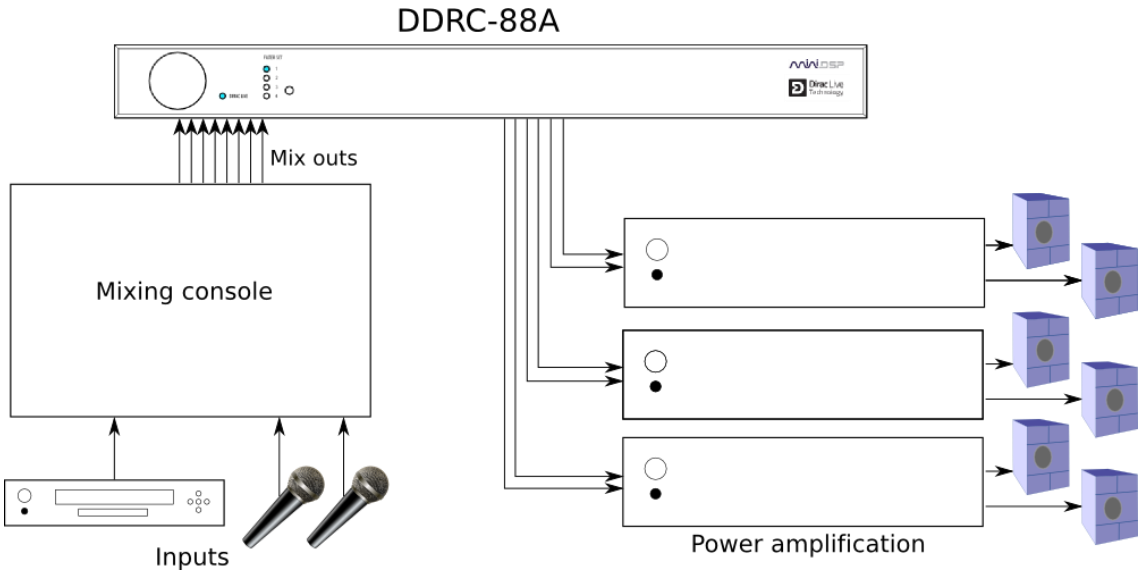
The *DDRC-88A* can be used anywhere an 8-channel room correction processor is required, such as in home theaters, recording and mastering studios, performance venues, places of worship, and so on.

1.1 Typical System Configurations

The *DDRC-88A* is inserted between line-level analog sources and power amplification. In home theater applications, signal to the *DDRC-88A* typically comes from an A/V receiver or A/V processor with one or more source devices, and the amplification is typically a multichannel power amplifier. A source such as a Blu-ray player with multichannel analog outputs can also be connected directly to the *DDRC-88A*.



In studio or sound reinforcement applications, the *DDRC-88A* is typically connected between a mixing console and power amplification, as shown below. Individual channels can be set for either full-range or subwoofer operation.



Computer connectivity is used to perform acoustic measurements and generate digital room correction filters. Up to four sets of correction filters can be stored on the *DDRC-88A* processor and recalled from the front panel or via an infrared remote. Once the processor is fully configured, the computer is no longer needed.

1.2 How Dirac Live® works

The miniDSP *DDRC-88A* audio processor includes Dirac Live®, a premium mixed-phase room correction technology. This technology is used not only in home stereo and home theater systems but also in cinemas, recording studios, and luxury cars.

As with any room correction system, Dirac Live® corrects the system's *magnitude response* (often referred to imprecisely as "frequency response"). In contrast to fully automated systems, Dirac Live® corrects the magnitude response towards a user-adjustable *target response*. The target response takes account of the natural frequency range of the loudspeaker system and the normal effects of loudspeaker dispersion on the measured *in-room* magnitude response.

In addition, Dirac Live® corrects the system's *impulse response*, which reflects how the system responds to a sharp transient such as a drumbeat. Reflections, diffraction, resonances, misaligned drivers, and so on, all combine to smear out the transient. An ideal loudspeaker has none of these, so correcting the impulse response makes the speaker in the room behave much more like that ideal loudspeaker. The impulse response is a critical factor for accurate sound-staging, clarity and bass reproduction. Dirac Live® employs a sophisticated analysis algorithm to make the optimal correction across the *whole* listening area, not just at a single point.

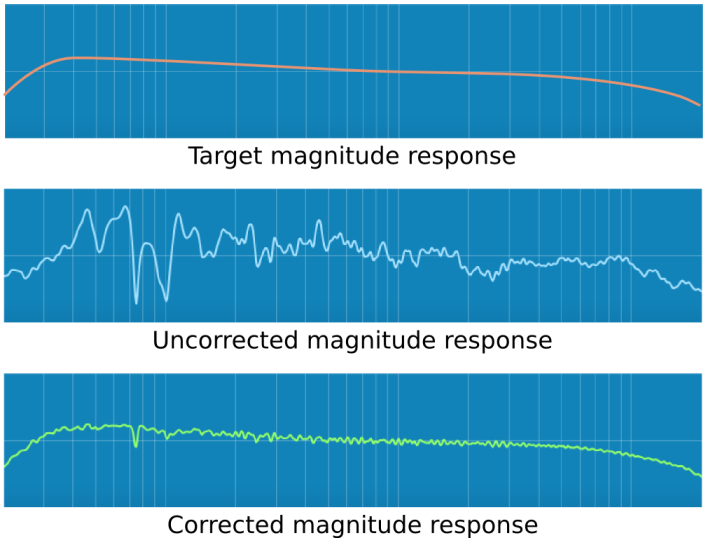


Illustration of Dirac Live® magnitude response correction

Dirac Live® accomplishes this using *mixed-phase filters* – filters that match a desired magnitude response *and* generate a customized impulse response. This contrasts with the *minimum-phase* and *linear-phase* filters that are commonly used in audio applications. While minimum-phase and linear-phase filters are relatively easy to design, they are tightly constrained in their impulse response characteristics – neither can make a desired change to the magnitude response independently of controlling the impulse response. In some cases, they may even make things worse.

Mixed-phase filters are more difficult to design, but the audible performance of Dirac Live® is due to its success in using mixed-phase filters to make the system response across the *whole* listening area more closely resemble that of an ideal speaker. The energy from the direct wave and from early reflections is optimally combined to arrive as a single wavefront to the listener. Late reflections are left largely untouched, being corrected only for their spectral coloration, as they contribute to a larger, more enveloping soundstage.

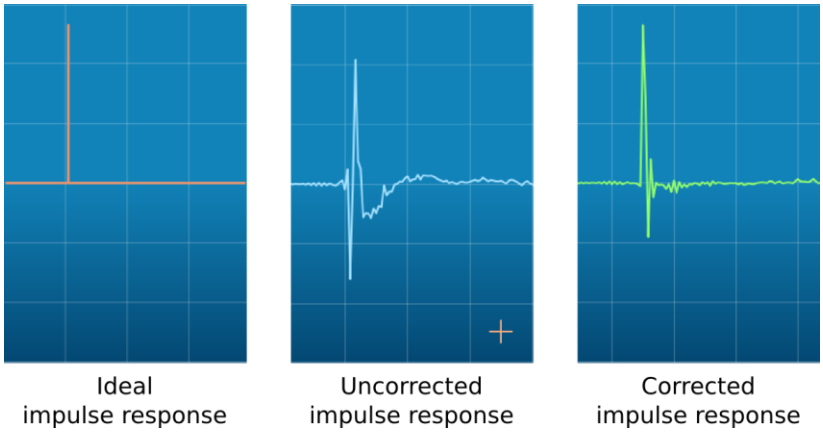
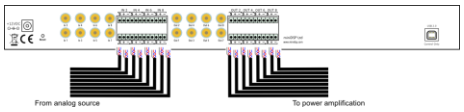


Illustration of Dirac Live® impulse response correction

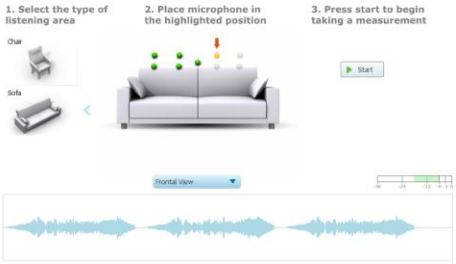
1.3 DDRC-88A / DIRAC LIVE® CONFIGURATION STEPS

The steps for configuring the *DDRC-88A* audio processor with Dirac Live® are summarized as follows:

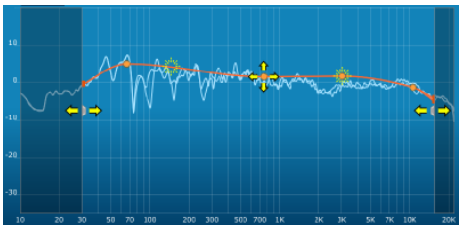
1. Connect the *DDRC-88A* audio processor into your system and install software. See Section 2, Installation and Setup.



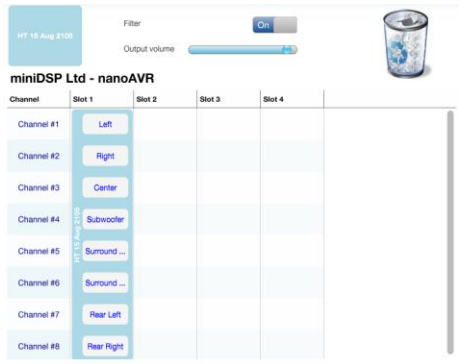
2. Run a series of acoustic measurements using the **Dirac Live Calibration Tool For miniDSP** program, to capture the acoustic behavior of your speakers and room. See Section 3, Acoustic Measurement.



3. Generate digital room correction filters that will be executed by the *DDRC-88A* processor. Up to four filter sets can be downloaded into the processor for easy real-time recall and auditioning. See Section 4, Filter Design.



4. Once the digital room correction filters are designed and downloaded, the computer can be disconnected for normal listening. See Section 5, Using the DDRC-88A audio processor.



2 INSTALLATION AND SETUP

2.1 SOFTWARE INSTALLATION AND LICENSE ACTIVATION

Please read and follow all steps in this section carefully. If you run into difficulties during installation and license activation, please contact miniDSP support via minidsp.desk.com.

2.1.1 Framework installation

Prior to installing the software, download and install the following frameworks. You will need to accept the license agreements in order to successfully complete the installation. If you haven't updated these recently, check that you have the latest versions prior to running the miniDSP install programs.

- Microsoft .NET framework (version 3.5 or later)
- Latest version of Adobe Flash
- Latest version of Adobe Air
- Microsoft Visual C++ 2010 Redistributable Package: for x86 (32-bit operating system) or x64 (64-bit operating system).

2.1.2 Software installation

When you receive notification that your order has shipped, your installation software downloads will be available at the *User Downloads* section of the miniDSP website:

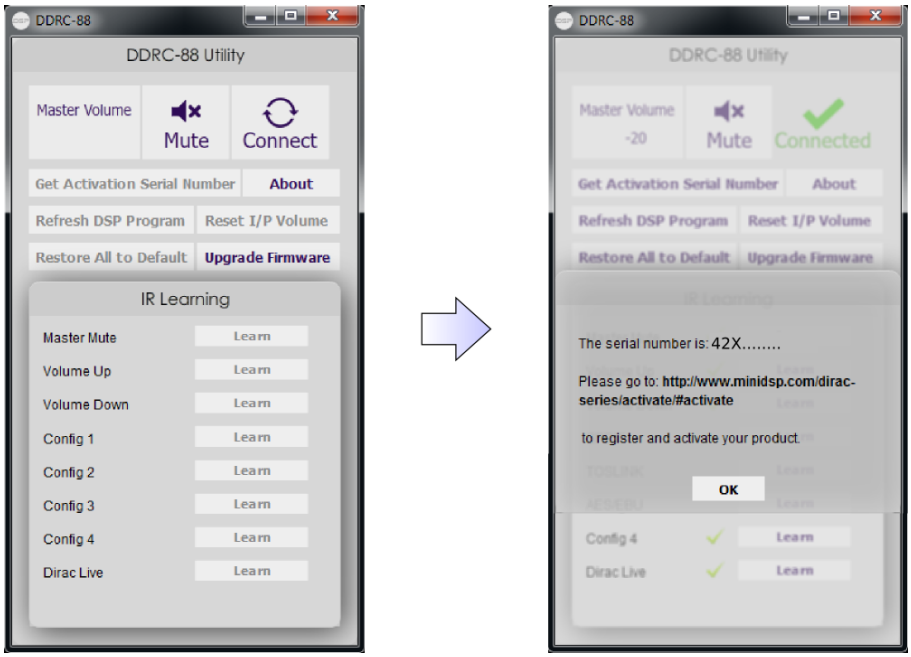
http://www.minidsp.com/userdownloads

Click on the *Dirac Series* heading and then on *DDRC-88A software*. Download two installation zip files:

- **Dirac Live Calibration Tools for DDRC-88A**. Unzip the downloaded file and double-click on the .exe installer file. Follow the on-screen instructions to complete installation. This will install the program **Dirac Live Calibration Tool** on your computer.
- **DDRC-88A Configuration Tool**. Unzip the downloaded file and double-click on the .exe installer file. Follow the on-screen instructions to complete installation. This will install the program **DDRC-88 Utility** on your computer.

2.1.3 License activation

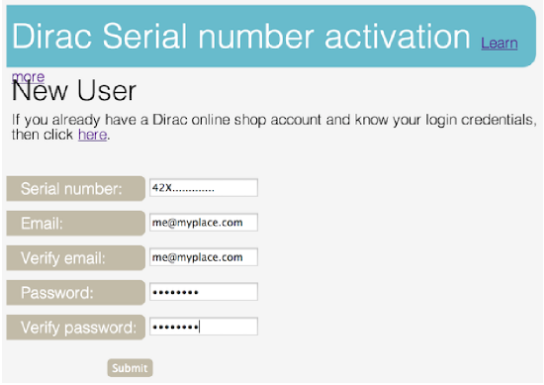
1. If you are running the **Dirac Live Calibration Tool** already, quit from that program. (You cannot run this program and the **DDRC-88 Utility** program at the same time.)
2. Start the **DDRC-88 Utility** program. It will appear as shown on the left below.
3. Connect your *DDRC-88A* processor to your computer via USB, then click on the **Connect** button. It will change to a green tick.
4. Click on **Get Activation Serial Number**. The program will get your unique serial number from the connected *DDRC-88A* and display it, as shown on the right below.



5. Using this serial number, activate your license on the miniDSP Dirac Live activation page:

   http://www.minidsp.com/dirac-series/activate/#activate

   If you have not activated a Dirac Live license before, use the **New User** screen on the left. If you have previously activated one or more Dirac Live licenses, use the screen on the right.



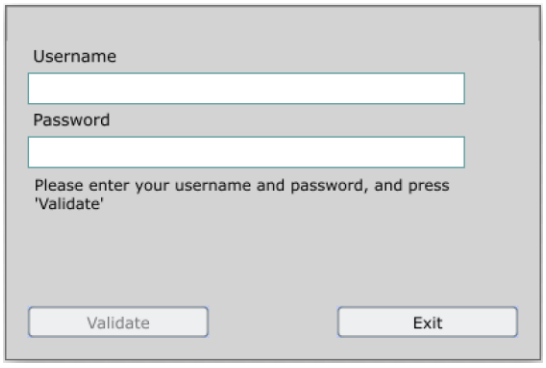
Dirac Serial number activation
Learn more
Input your Dirac online shop account and password to activate your Dirac product serial number
If you do not have an Dirac online shop account, click here
Serial number:
42X.....
Email:
me@myplace.com
Password:
Submit

Notes:

1. The serial number that must be entered in the activation screen is **not** the serial number printed on the hardware unit. A unique activation serial number for Dirac Live is programmed into the firmware of each unit and can only be accessed with the **DDRC-88 Utility** program (step 4 above).
2. The email address used during license activation is not related to your user account on miniDSP.com. To activate and validate a Dirac Live license, you must create a separate account on the Dirac server using the form shown at left in step 5 above.
3. If you purchase more than one *DDRC-88A* or *Dirac Series* processor, each will need to be activated with its unique serial number. Use the form shown on the right in step 5 above and enter the activation serial number of each additional processor, and they will in turn be added to your license.

2.1.4 License validation

Start the **Dirac Live Calibration Tool For miniDSP** program. It will ask you to validate your software license. Enter the email address and password that you used when activating your license, and then click on **Validate**:

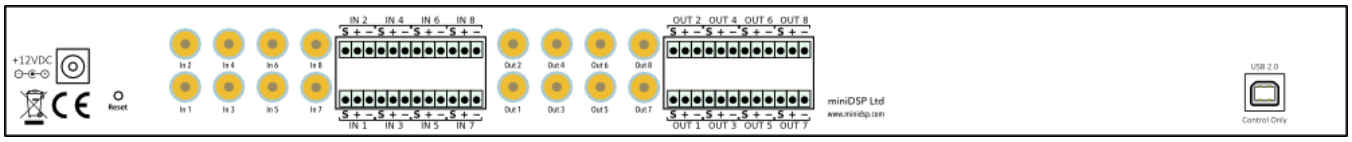


You will need to be connected to the Internet to validate your license. Any active firewalls will need to have HTTP (normal web traffic) enabled.

The **Username** that you enter in the validation screen must be the *email address* that you used in the activation screen in step 5 on page 12. It is **not** your miniDSP website username.

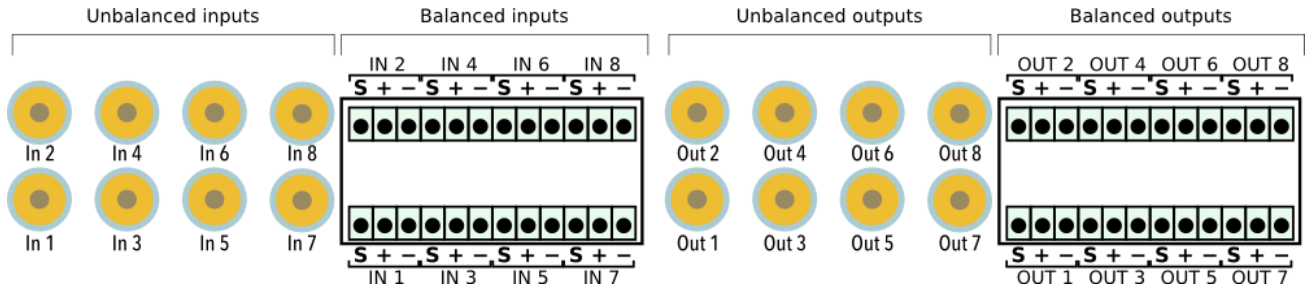
2.2 HARDWARE CONNECTIVITY

All connections to the *DDRC-88A* are made on the rear panel.



2.2.1 Analog input and output

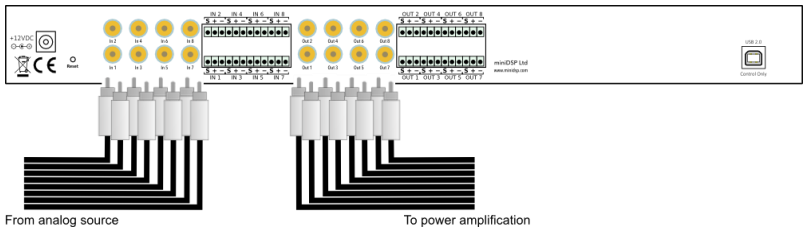
Up to eight channels can be connected to the *DDRC-88A*. Be sure to take careful note of the channel numbering:


Unbalanced inputs
Balanced inputs
Unbalanced outputs
Balanced outputs
IN 2 IN 4 IN 6 IN 8
In 2 In 4 In 6 In 8
In 1 In 3 In 5 In 7
IN 1 IN 3 IN 5 IN 7
Out 2 Out 4 Out 6 Out 8
Out 1 Out 3 Out 5 Out 7
OUT 2 OUT 4 OUT 6 OUT 8
OUT 1 OUT 3 OUT 5 OUT 7

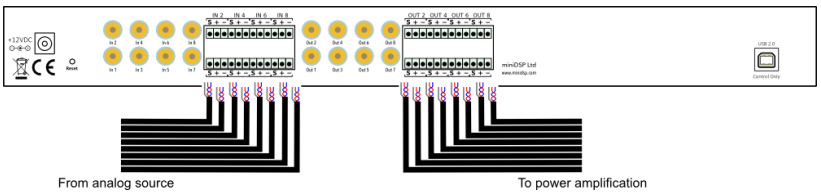
On the **input** side, it is *very* strongly recommended that all connections be of the same type – i.e. balanced or unbalanced. The Dirac Live algorithm equalizes levels on all channels, and a mixture of balanced and unbalanced connections on the *input* side will require special gain adjustment steps after Dirac Live calibration.

On the **output** side, a mixture of balanced and unbalanced connections can be used if desired. Doing so may necessitate additional work to check and adjust system gain structure. For more information on gain structure, see Gain structure starting on page 38.

Unbalanced connections are made directly to the RCA jacks.

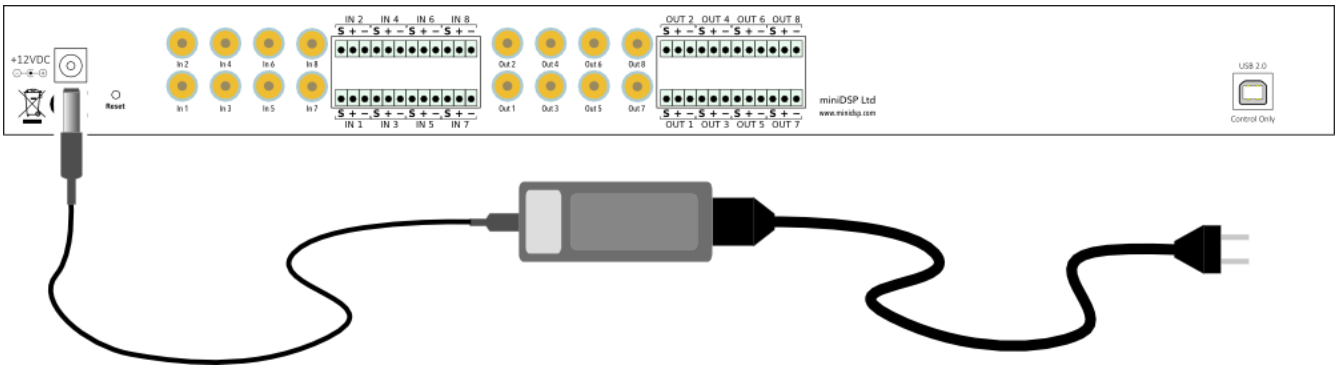


Balanced connections are made by connecting bare wire ends to the push-in Phoenix terminal blocks. For advice on using these, see Balanced wiring tips on page 48.



2.2.2 DC Power

Fit the supplied IEC cable to the 12 VDC power supply. Plug the AC mains plug into the power outlet, and then plus the DC connector into the **+12VDC** socket on the rear panel of the *DDRC-88A*.



Apply power to the *DDRC-88A* only after all analog input and output connections have been made. The DDRC-88A uses little power and can be left powered on.

If powered on and off, the following turn-on/turn-off sequence is recommended:

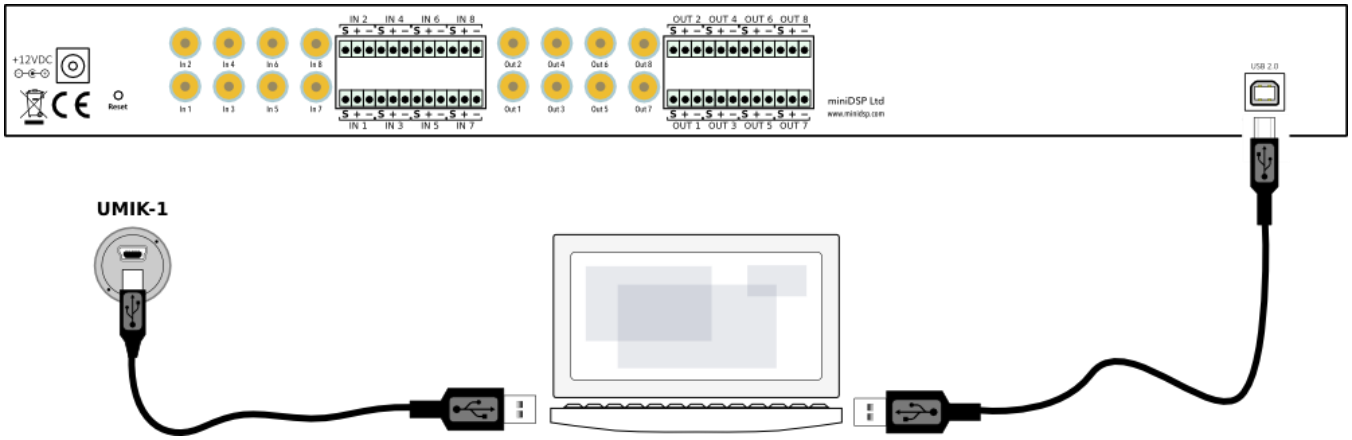
- On: Power on line-level equipment, including the *DDRC-88A*, then turn on power amplification.
- Off: Turn power amplification off, then power off line-level equipment, including the *DDRC-88A*.

2.2.3 USB

To configure the *DDRC-88A* using **Dirac Live Calibration Tool for miniDSP**:

- Connect the USB port of the *DDRC-88A* to a USB 2.0 port on your computer using the supplied cable.
- Connect a miniDSP UMIK-1 to a second USB port on your computer.

Note: the miniDSP UMIK-1 is the only measurement microphone that can be used with the *DDRC-88A* and **Dirac Live Calibration Tool for miniDSP**.

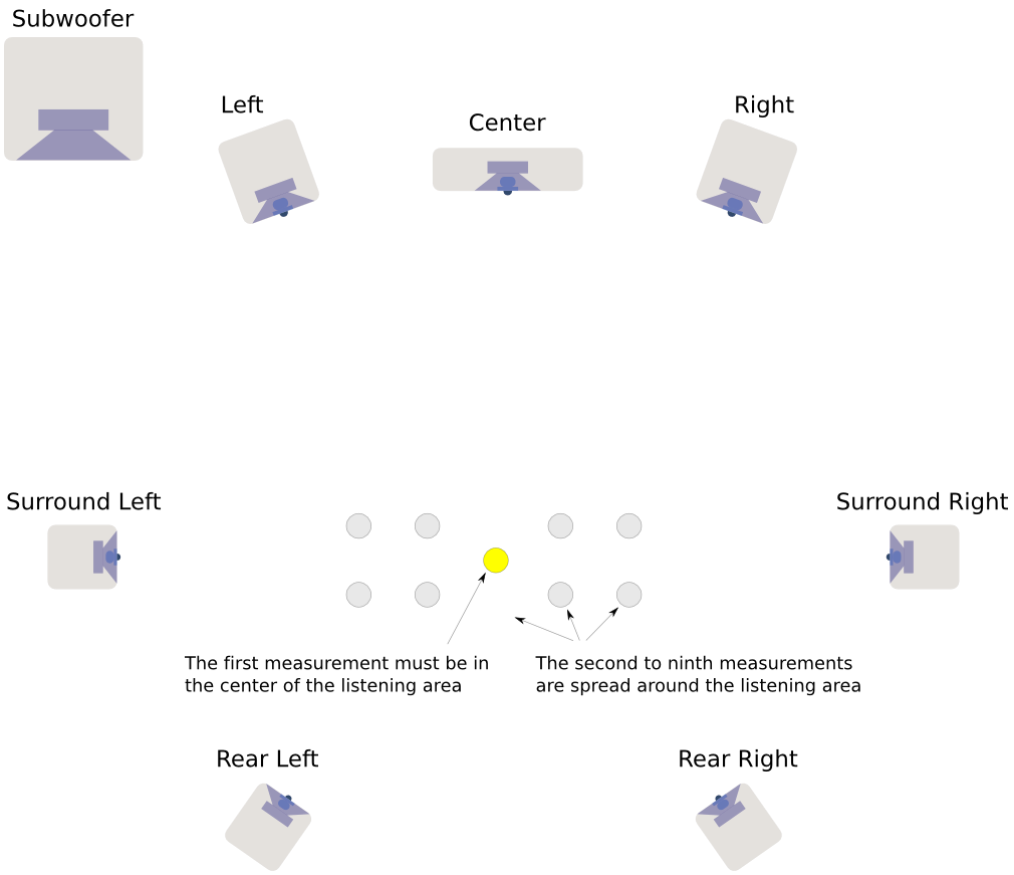


3 ACOUSTIC MEASUREMENT

The **Dirac Live Calibration Tool For miniDSP** uses a set of measurements made in your listening room to gather all the acoustical information about your room and speakers that it needs to calculate the correction filters. The measurements are made using the *DDRC-88A* processor and a miniDSP UMIK-1 measurement microphone (must be purchased separately).

3.1 LOUDSPEAKER AND MICROPHONE POSITIONING

Prior to performing acoustic measurements, loudspeaker and subwoofer positioning should be optimized. In a home theater setting, the location of the subwoofer within the room will have a large impact on the smoothness of bass response. With Dirac Live®, you have more freedom with loudspeaker and subwoofer placement, but the best result will still be achieved if optimal placement is used together with Dirac Live ®.



Typical home theater measurement setup

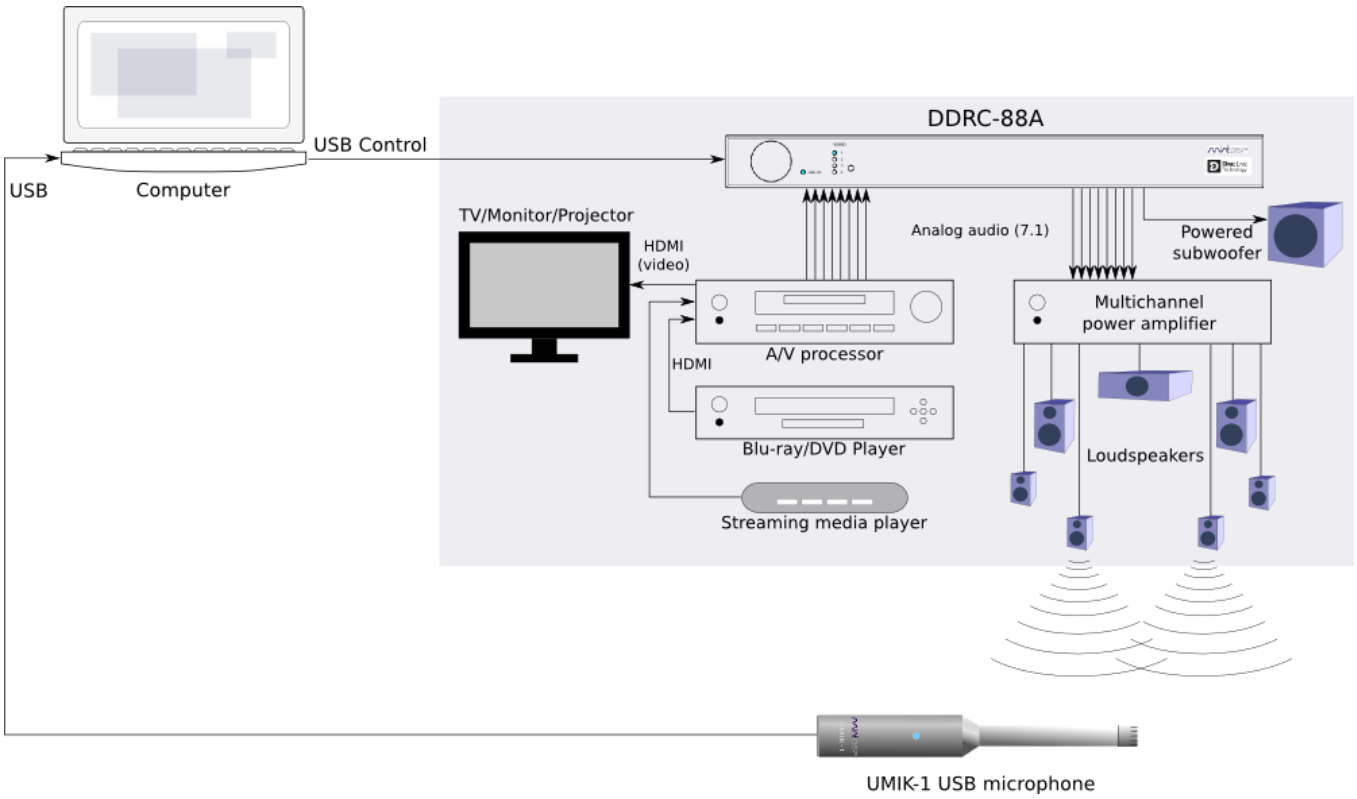
A total of nine measurements are needed, with the microphone located in different positions in the room and pointed vertically (that is, at the floor or ceiling). The first measurement must be taken at the central location of the listening area, as this location sets the levels and delays of each speaker. Eight more measurements are then taken at locations spread around the listening area and at different heights from the floor.

3.2 PREPARING FOR ACOUSTIC MEASUREMENT

3.2.1 Connections and microphone placement

The figure below shows a typical connection diagram for performing acoustic measurement. No changes to the audio connections are needed. Simply:

1. Connect the supplied USB (type A to type B) cable from the *DDRC-88A* to a USB port on the computer.
2. Connect a USB cable (type A to mini type B) from the UMIK-1 to a USB port on the computer.



Place the UMIK-1 microphone into a microphone stand and position the computer and cabling so that there is enough freedom of movement to move the microphone into the needed locations. A small tripod stand is supplied with the UMIK-1, but a larger stand with boom arm can be used if desired. If necessary a USB extension (up to a total USB cable length of 5 meters) can be used. In larger spaces, an active USB repeater may be needed. We recommend that the microphone be oriented vertically (pointed at the floor or ceiling) and the "90 degree" calibration file used (see Mic Config tab below).

3.2.2 Subwoofer settings

We recommend that the low pass filter of the subwoofer be disabled if possible, or set to its highest frequency if it cannot be disabled. Any EQ on the subwoofer should be disabled or set "flat." High pass filters used to protect the driver from over-excursion should be left in place.

If the subwoofer's low pass filter is an important part of the overall bass management in the system, it can be re-enabled after completing Dirac Live calibration and loading correction filters into the *DDRC-88A*.

3.3 CONFIGURING FOR MEASUREMENT

Start **Dirac Live Calibration Tool For miniDSP (DLCT)**.

Be sure to quit the **DDRC-88 Utility** program before starting **Dirac Live Calibration Tool for miniDSP**. Running the two programs at the same time will result in communication conflicts and odd behavior.

Logo and status/progress bar
Click to show and hide help pane
Main display area for each selection tab
Sound system
Mic config
Output & Levels
Measurements
Filter Design
Export
Screen selection tabs
Load and save a project
Back to previous / proceed to next screen
Load...
Save...
Back
Proceed
Output selection and level adjustments
1. Make sure that the "Output volume" slider is set to about 20% of max. If the slider is disabled, adjust the physical level control on your hardware device.
2. Pressing the "Test" button on any channel will start playback of a test noise which is used to verify that each output is connected to the correct speaker. By pressing the test buttons, check that the "Channel names" given in the boxes to the left match the physical speakers of your audio system. If the noise of a channel appears in the wrong speaker, use the "Output channel" dropdown menu to select the correct speaker.
3. Set the volume on your amplifier as you would during normal listening and gradually turn up the "Output level" slider until you reach normal/high listening level.
4. Place the microphone in the center of the listening area and play the test noise on every channel.
5. If you are using a microphone pre-amplifier with a gain control, set

Logo and status progress bar

This area shows a progress bar with current status when the program is performing calculations. If the program seems unresponsive at any time, check the status here.

Screen selection tabs

Each tab selects the information shown in the main display area. These are generally worked through in order, from top to bottom. This section covers the first four tabs; the final two are covered in Filter Design.

Load and save a project

A set of measurements can be saved to a file and reloaded at a later time. See Saving and loading projects.

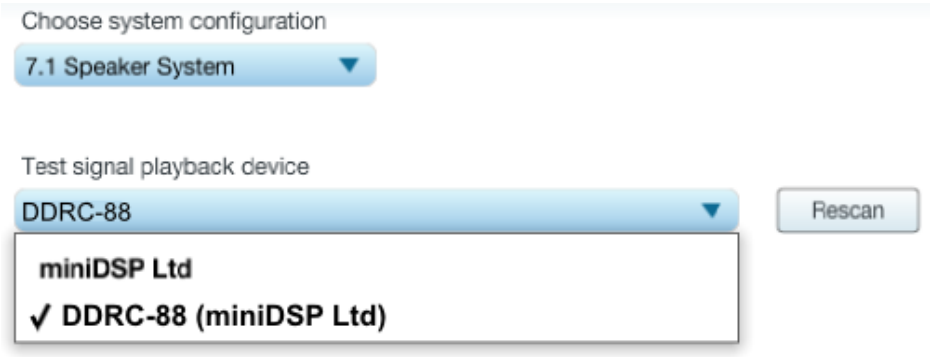
Back / Next

Use these two buttons to advance to the next tab when each is complete, or to go back to the previous tab to make alterations. The tabs at the left can also be clicked on directly.

Help open/close

Click on the small Help divider at the right of the screen to open a pane with help on the currently selected tab. Click on the divider again to close the help pane.

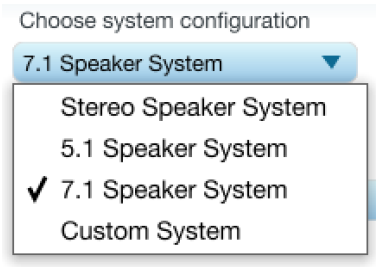
3.3.1 Sound System tab



On the **Sound System** tab, set the following parameters.

Choose system configuration

Use the dropdown menu to select your system configuration.



For multi-channel surround-sound, use **5.1 Speaker System** or **7.1 Speaker System** if it is desired that Dirac Live calibrate for a 10 dB "LFE alignment gain" to the subwoofer channel. If the calibration for LFE alignment gain is not required (that is, external equipment is providing this gain) or if a different speaker configuration is required, use **Custom System**. (See Custom System configuration below for further information.)

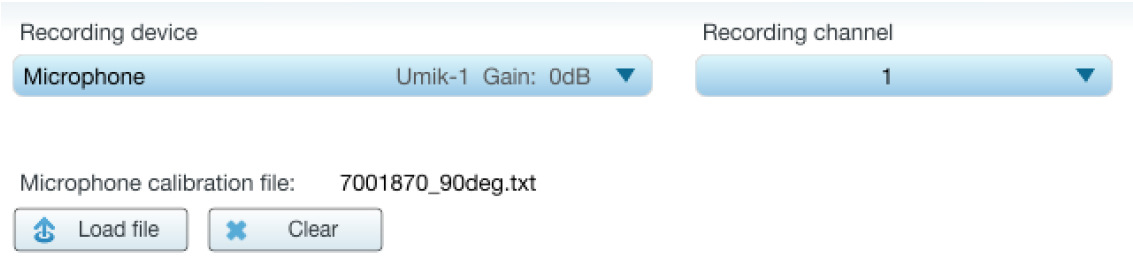
Test signal playback device

Preset to **DDRC-88 (miniDSP Ltd)**. This will ensure that test signals are sent into your audio system via the *DDRC-88A* processor.

If the entry for **DDRC-88** is not showing, check that your *DDRC-88A* processor is connected via USB and powered on, click the **Rescan** button, and then use the drop-down menu to select **DDRC-88**.

Once you have verified that this tab is correct, click the **Proceed** button.

3.3.2 Mic Config tab



On the **Mic Config** tab, set the following parameters.

Recording device

Preset to the **UMIK-1**.

If the UMIK-1 is not showing, ensure that the UMIK-1 is connected securely to the computer via USB, then go back to the **Sound System** tab and click on **Rescan**. Then use the drop-down menu to select the "Microphone" item underneath "UMIK-1".

Recording channel

Select **1** from the drop-down menu.

Microphone calibration file

Each UMIK-1 measurement microphone is individually calibrated to ensure accuracy. To download the unique calibration file for your microphone, go to the UMIK-1 page and enter your microphone's serial number. It is in the form xxx-yyyy and labelled on the microphone. Ensure that you download both the regular calibration file and the "90-degree" calibration file. (The latter is generated specifically for use with miniDSP's multi-channel Dirac Live® processors such as the *DDRC-88A* and the *nanoAVR DL*.)

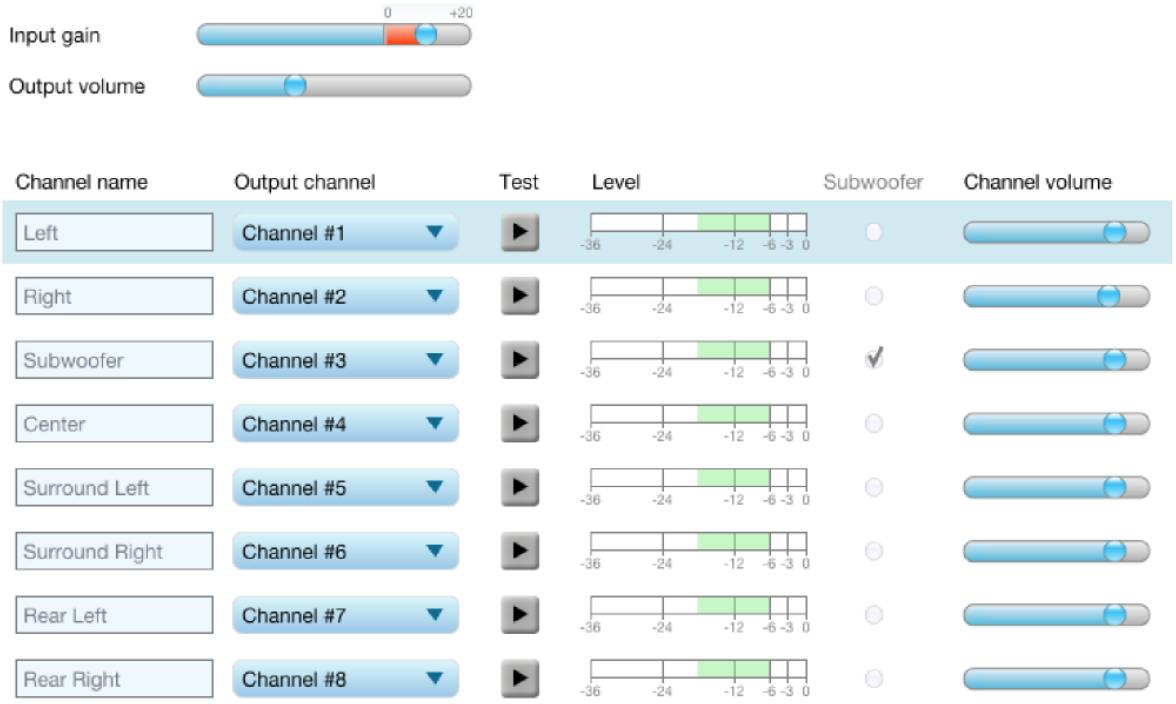
Then click on the **Load File** button and select your calibration file.

For home theater applications, it is best to use the 90-degree calibration file as this is created specifically for the vertical microphone orientation. This file is downloaded with the suffix "_90deg" in the file name.

Once you have verified that this tab is correct, click the **Proceed** button.

3.3.3 Output & Levels tab

On the **Output & Levels** tab, set **Output volume** quite low. If you have another volume control "down-stream" of the *DDRC-88A*, set it about halfway and increase it later if needed.



Click on the **Test** button for the left channel and gradually increase the output volume until it is at a moderate level, such that your voice would have to be raised to converse with someone sitting next to you.

Increase the **Input gain** slider until the blue bar on the level meter reaches up into the green section:

Click again on the **Test** button for the left channel to stop the test signal. Then click on the **Test** button for each of the remaining channels. If any channel is not in the green zone, use its **Channel volume** slider to adjust the volume of the test signal for that channel. (Some readjustment of **Input gain** and **Output volume** to get all channels "green" may be required.)

If large adjustments are required to the **Channel volume** sliders, you may have a gain structure problem. In that case, please refer to the section Optimizing gain structure starting on page 42.

When done, click the **Proceed** button.

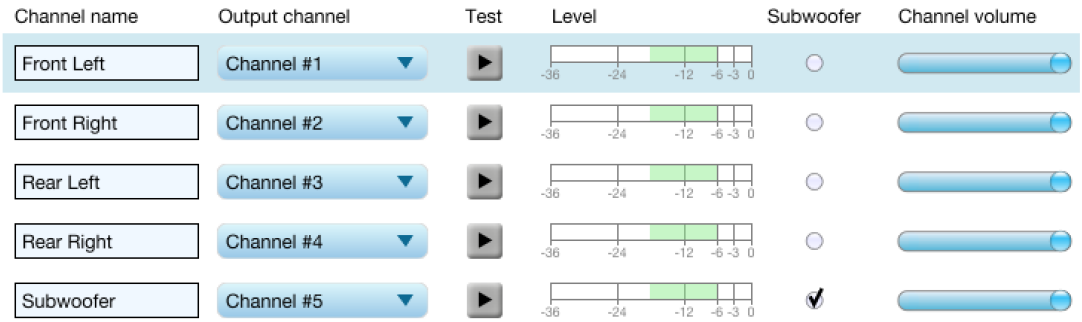
3.3.4 Custom System configuration

On the **Sound System** tab, choose the **Custom System** configuration if any of the following apply:

- Your system does not fit any of the three predefined configurations (Stereo, 5.1, 7.1).
- You want to use a different channel mapping than the default.
- You do not want the *DDRC-88A* to calibrate for a 10 dB LFE alignment gain on the subwoofer channel.

After choosing **Custom System**, you will need to select the number of channels that you want to use.

When you get to the **Output & Levels** tab, it will show controls for the number of channels that you selected:



Channel name

Type any name you like for each channel.

Output channel

By default, each input channel maps to the same numbered output channel (input channel 1 to output channel 1, and so on). The dropdown selectors can be used to change this mapping. Note that DLCT will not let you assign more than one output channel to each input channel.

If your custom system configuration has less than eight channels, inputs and outputs are assigned in sequential order. For example, for a 4.1 channel system, connect to inputs and outputs 1 through 5.

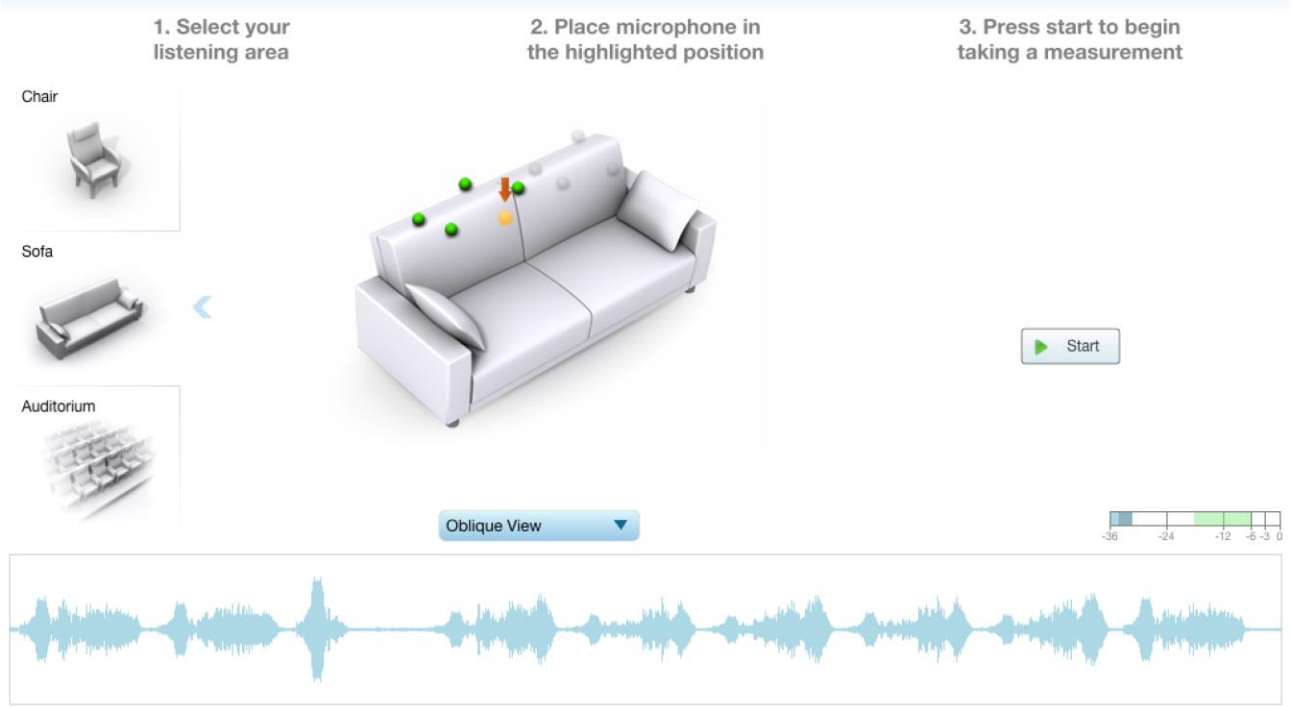
Subwoofer

The subwoofer checkbox tells the Dirac Live analysis algorithm to use a different method to detect the impulse on that channel, which in turn affects the delay that will be assigned to that channel. This is needed because of the limited frequency response of the subwoofer.

Reminder: in **Custom System** configuration, the *DDRC-88A* does not calibrate for a 10 dB LFE alignment gain on the subwoofer channel. This gain, if required, must therefore be provided by external equipment. If in doubt, check the specifications of your equipment and refer to LFE alignment gain on page 44 for more information.

3.4 RUNNING THE MEASUREMENTS

Acoustic measurements are performed on the **Measurements** tab.

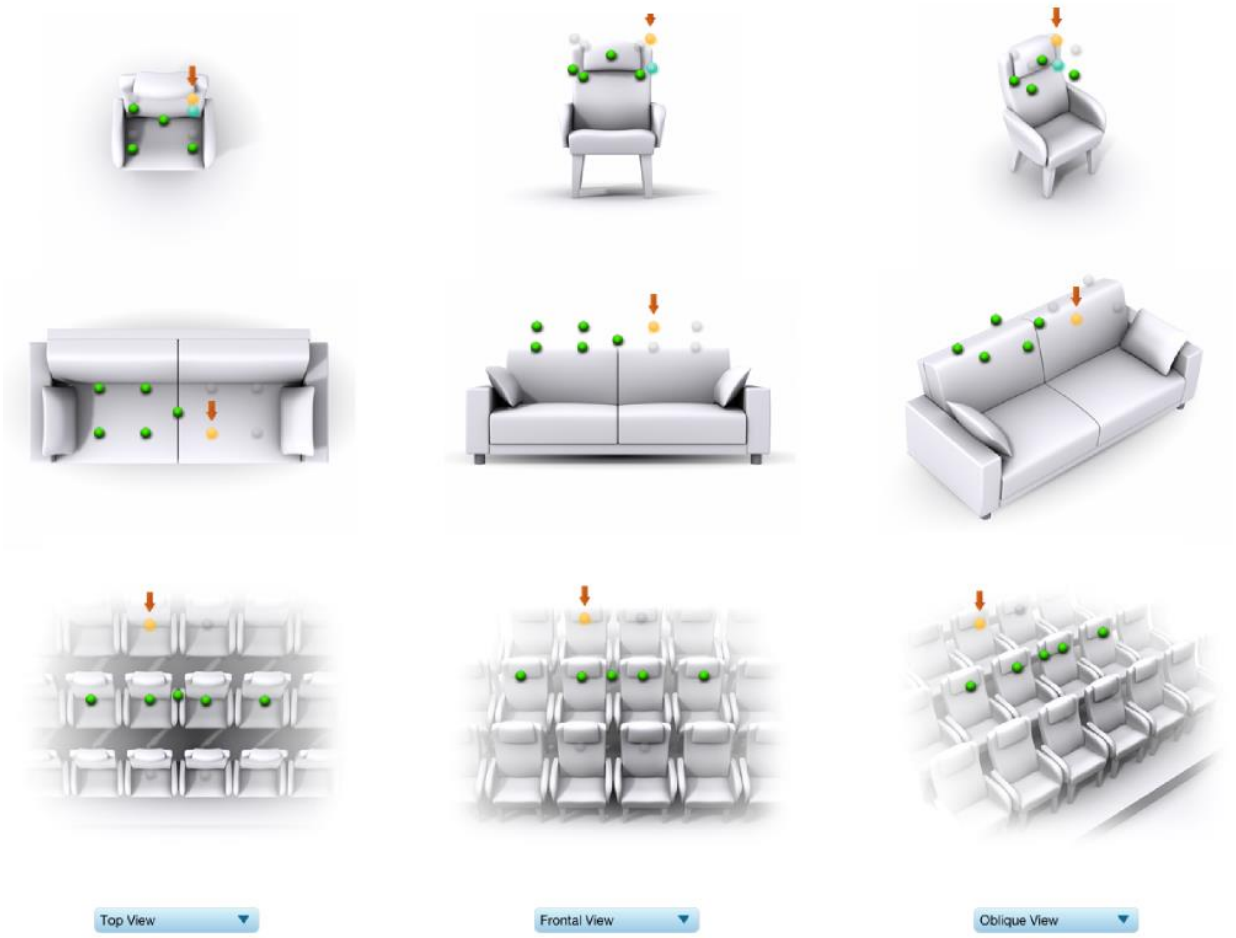


Measurements should be performed under good conditions if possible. While the measurement technique used by Dirac Live is quite robust, low-frequency noise (traffic, machinery, aircraft, storms) in particular can adversely affect measurement accuracy. A high level of ambient noise can degrade signal to noise ratio and prevent the algorithm from analyzing the test sweep signal properly. Minimize the effect of any external noise, ensure that measurement signal levels are adequate, and/or choose a suitable time for performing measurements.

3.4.1 Listening environment

The **Measurements** tab presents three different listening environments as a visual guide to positioning the microphone for each of the nine measurements: **Chair**, for a single listening seat; **Sofa**, for multiple listening seats; and **Auditorium**, for a large dedicated home theater or larger venue with staggered seating. Use the icons at the left of the screen to select the listening environment.

The center of the screen contains a pictorial representation of the selected listening environment, with dots marking the recommended microphone locations. Completed measurements are shown in green, while the next measurement to be done is highlighted in yellow and has a red arrow marker pointing to it. A drop-down menu underneath selects three different views, which should be used to help you place the microphone in a suitable location.



While the visual guide indicates a suitable set of microphone locations, these locations can be varied to suit individual circumstances. It is, however, imperative that the first measurement is taken in the center of the listening area, as this measurement is used to set the levels and delays of each channel. The subsequent eight measurements should be well spread out over the entire listening area so that Dirac Live can acquire a good set of measurements that capture the acoustic behavior of the room. Placing all microphone locations too close to each other may result in "over-correction" that will sound dry and dull.

For example, if using the **Chair** listening environment, spread the microphone positions over a circle with a diameter of a meter (three feet). Vary the height of the microphone up and down by 30 cm (one foot) from the initial position. If using the **Sofa** or **Auditorium** listening environment, again spread the measurement locations over the full listening area and vary microphone height by a significant amount.

A different set of locations other than those indicated by the visual guide and the above guidelines can be used if necessary. The important thing is to ensure that the measurement locations are spread out over the whole listening area and that the microphone is moved a sufficient distance vertically as well as horizontally.

In some cases, such as when the listening area is very close to the loudspeakers or the loudspeakers have a very narrow dispersion pattern, the size and in particular the height of the measurement area can be reduced, to avoid discrepancies caused by varying output response from the speakers themselves.

3.4.2 Executing measurements

With the microphone in place at the central location and pointed vertically (that is, towards the ceiling or floor), click on the **Start** button. The *DDRC-88A* will generate a test signal, audible as a frequency sweep through the left speaker, then the right, and so on through all channels. Finally, the frequency sweep plays through the left speaker again.

While the measurement proceeds, the time-domain response graph of the captured audio signal is displayed at the bottom of the measurement tab. (This graph is related to the magnitude response but is not the same display. Its purpose is to verify that the recorded signal level is in a suitable range.)

After completion of the measurement sweeps, the status bar will update with a progress indicator as the program performs calculations on the measurement. If the measurement was successfully captured, the red arrow marker will advance to the next location to be measured.

If the program indicates that the measurement was not successful, you will need to take corrective action. The most common errors are related to signal level:

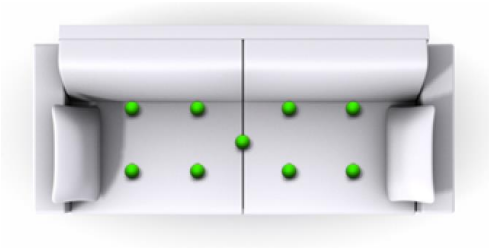
- The measurement signal is too low to ensure a clean capture.
- The measurement signal is too high and the audio signal has exceeded the maximum level (clipping). This is shown in red on the signal graph.

In either of the above cases, go back to the **Output & Levels** tab and adjust **Output volume**, **Input gain**, or the **Channel volume** slider for the channel that caused the problem. Then re-run the measurement. (You do not need to redo the measurements you have already successfully completed.)

3.4.3 Completing the measurements

After each successful measurement, the location marker (red arrow) will advance to the next location. Move the microphone to that location, using the three views (top, front, oblique) as a guide to positioning it. Then click on **Start** again. Repeat this process until all nine locations have been successfully measured.

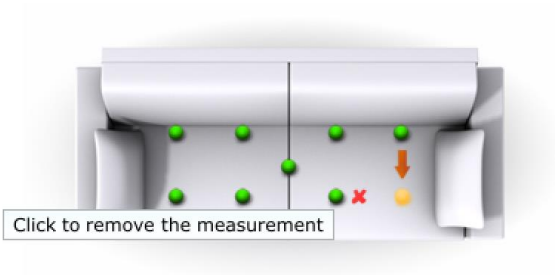
Note: it is good practice to save the project periodically while performing measurements (see Saving and loading projects below).



3.4.4 Viewing and redoing measurements

Click on the green dot for any completed measurement to display its measured time-domain response graph.

After clicking on a green dot, a small red "X" will appear next it. Click on the "X" to delete the measurement. The status bar will indicate that the program is recalculating parameters.



To redo a measurement, delete it, move the microphone to the appropriate location, and click on **Start**. Note: if more than one measurement is deleted, the marker will move to the lowest-numbered one.

It is important that all nine measurements are completed in order to ensure best results from the optimization algorithm. Being patient and thorough will pay audible dividends!

Once all nine measurements have been successfully completed, click the **Proceed** button.

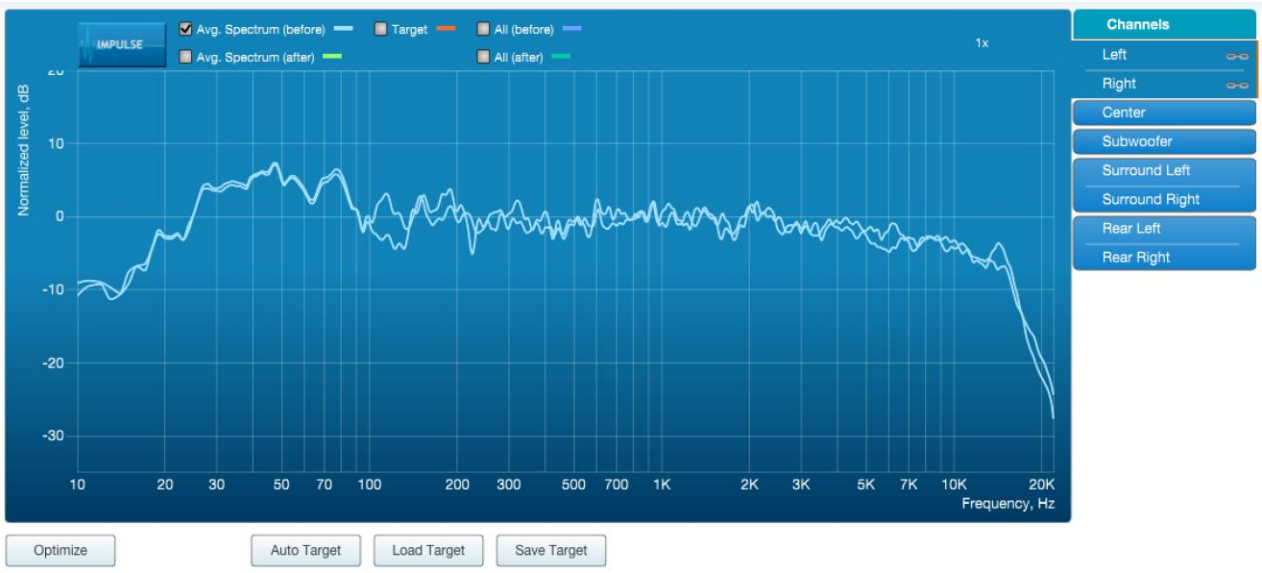
3.5 SAVING AND LOADING PROJECTS

Each set of measurements and the associated configuration settings are called a *project*. The project should be saved at regular intervals by clicking on the **Save** button. The default location for project files is **C:\Users\[User]\Documents\MiniDSP\nanoAVR\Projects**, but you can store them in any location in your file system.

A project can be reloaded at any time by clicking on the **Load** button. This enables you to generate new correction filters for different target curves at a later date, or to redo any of the measurements. (Note: if you wish to change between the **Chair, Sofa**, or **Auditorium** listening environments, you will need to start a new project.)

4 FILTER DESIGN

The **Filter Design** tab shows sets of graphs for the various channels. Click on the tabs at the right to display the response graphs for different sets of channels (left and right, center, subwoofer, and surrounds, in the case of 5.1 and 7.1 systems). For each set of graphs, a number of variants can individually be turned on and off with the checkboxes above the graphs.



Avg. spectrum (before)

The average of the measured magnitude responses. These plots are shown in light blue.

Avg. spectrum (after)

The predicted average magnitude response after correction. These plots are shown in green, and can be viewed only after filters have been generated with the **Optimize** button.

Target

The target curve – that is, the desired in-room magnitude response. This curve is user-adjustable so you can fine-tune it to best suit your speakers, room, and preferences. See Designing your target curve below.

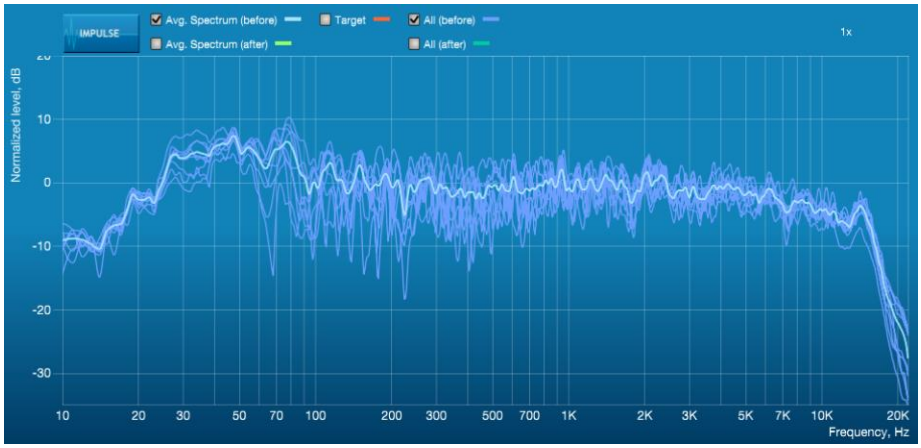
All (before)

All of the measured magnitude responses. These plots are shown dark blue.

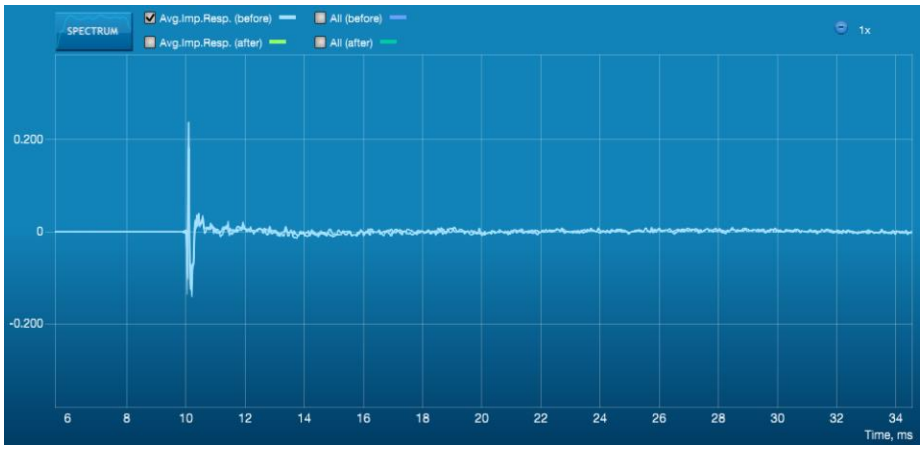
All (after)

All of the predicted magnitude responses after correction. These plots are shown in dark green, and can be viewed only after filters have been generated with the **Optimize** button.

The graphs showing all nine measurements are useful for seeing how much variation there is across the listening area:



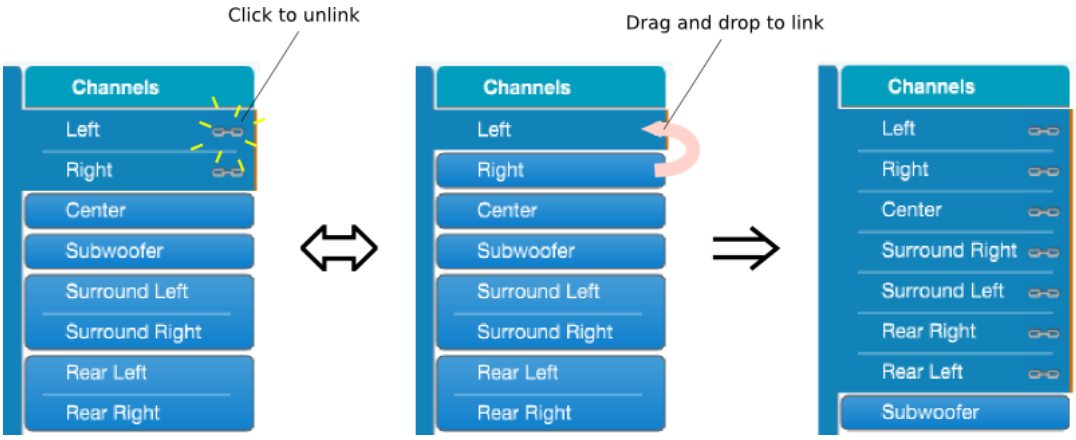
To display the impulse response instead of the magnitude response, click on the **Impulse** button at the top left of the display. All nine individual impulse responses can be shown as well as the average response. The predicted responses after correction can be viewed after filters are generated with the **Optimize** button (see Generating correction filters below).



To return to the magnitude response, click on the **Spectrum** button.

4.1 Working with graphs

Initially, the left and right front channels are shown. Some channels are linked together, as indicated by the small "chain" icons on the tabs at the right of the graph. When channels are linked, their graphs display together, and they share the same target curve. By default, the front left and right, surround left and right, and rear left and right channels are linked.

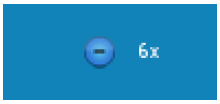


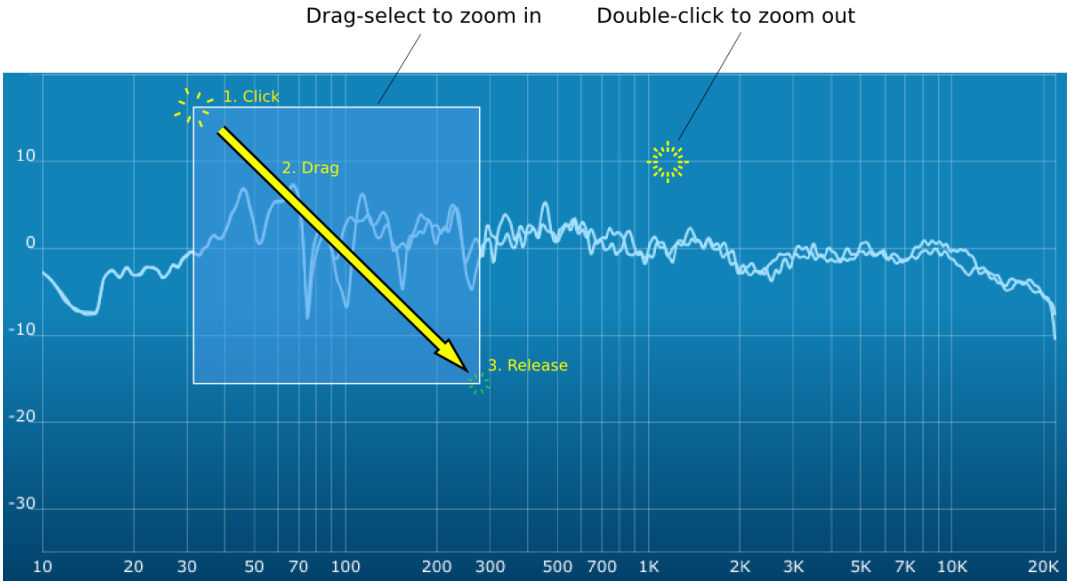
To unlink a channel, click on its chain icon. It will then be unlinked from the other channels. To link it to another channel or groups of channels, drag its tab on top of the channel or group of channels that you want it linked to.

Initially, you may wish to link all speaker channels together, as shown at right in the diagram above, as this will make it easier to experiment with target curves. Once you are up and running with target curves and filter sets, you can experiment with different target curves for different channel groups.

The response graphs can be viewed at a larger scale. To zoom in and out:

- Drag-select a region of the graph to zoom in on it. (Click the left button, move the mouse while holding the button, release the button.) You can then drag-select a region again to zoom in further.
- Double-click on the graph to zoom back out to the previous zoom level, or click on the small "–" sign next to the zoom indicator at the top right of the display.





4.2 DESIGNING YOUR TARGET CURVE

The *target curve* is the desired in-room magnitude response with the *DDRC-88A* processor performing digital room correction.

4.2.1 The Auto Target

When first viewing the **Filter Design** tab, an estimated target curve suitable for your speakers is shown as the red curve. This calculated target curve can be restored at any time by clicking on the **Auto Target** button.

Note: restoring the auto target will erase the current target curve. If you wish to keep it, you can save it to a file – see Saving and loading target curves below.

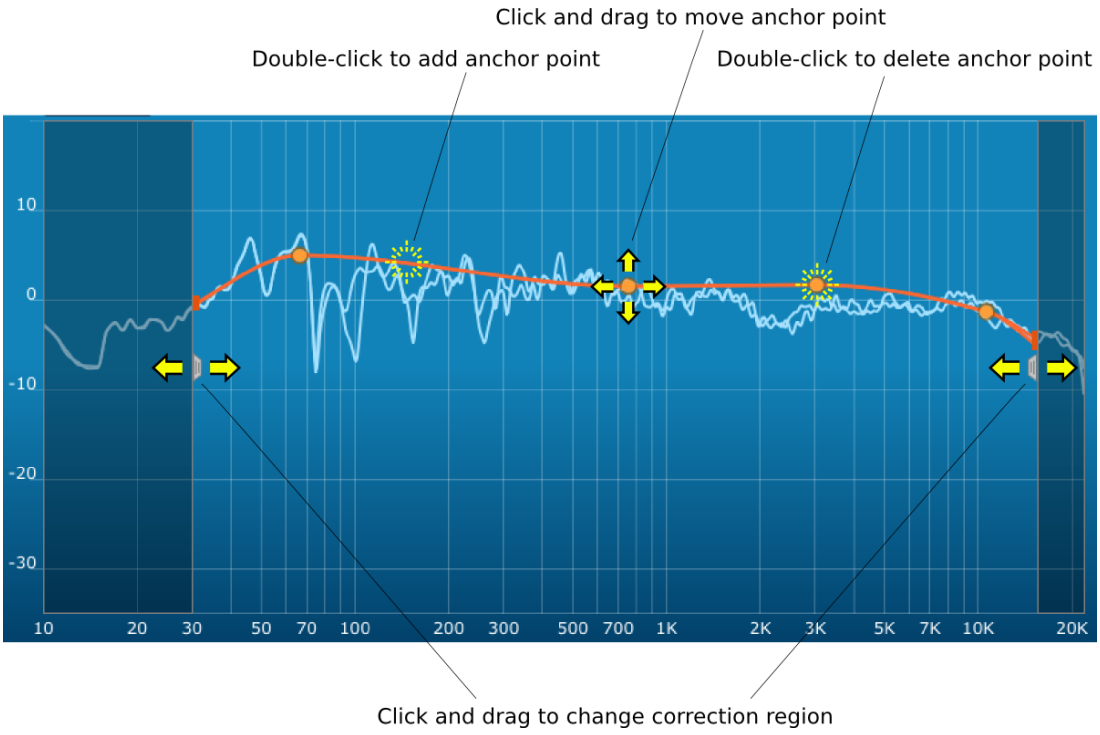
4.2.2 Editing the target curve

You can edit the target curve to produce any desired magnitude response. This is done with the use of *anchor points*, shown as orange dots on the curve:

- Drag an anchor point to move it.
- Double-click on the target curve to add an anchor point.
- Double-click on an anchor point to delete it.

The regions to the left and right of the response graphs that are shaded in a darker color are excluded from magnitude response correction. You can adjust the frequency range for your system and preferences. For example, low-frequency noise (traffic, machinery) may be present in some environments, so it is best to adjust the frequency range to exclude these frequencies from the correction. Or, you may be happy with the in-room response at higher frequencies, so you can set the frequency region to limit correction to the modal region (up to 300 Hz, in a typical room).

To alter the region of correction, drag the grey handles on either side of the graph. Note that you can't drag these handles over an anchor point, so you may need to move or delete an anchor point that is "in the way."



If channels are linked, the same target curve is used for that group of linked channels. To create a separate target curve for a single channel, unlink it as described above in Working with graphs.

4.2.3 Guidelines for target curve design

Care should be taken to create a target curve that works well with your speakers and room, as well as suiting your personal preferences. Small changes to the target curve can have significant effects on the tonal quality of the system, so it is important that you experiment with different target curves to find the optimum.

If you initially don't achieve a satisfactory result, please ensure that you have spread your measurements over a sufficiently large area and with sufficient variation in height. The following guidelines will help you understand how to adjust your target curve.

Low-frequency extension and boost

All loudspeakers have a natural low-frequency roll off. Setting the target curve to boost the region below the speaker's natural roll off frequency *may* result in overdriving the speakers, especially with smaller loudspeakers and depending on your listening habits. As a general rule, a home theater system should use bass management to direct low frequency content to the subwoofer.

The auto-target estimates the low-frequency roll-off and curve. You should determine by listening whether this estimate is suitable for your system, and adjust the target curve accordingly.

High-frequency "tilt"

The target curve is the desired measured response of loudspeakers *in a room,* In contrast to measurements made of a loudspeaker during its design under anechoic (measured in free space) conditions. While high-quality loudspeakers are usually designed for a flat on-axis anechoic response, these same speakers when placed into a listening room will tend to have a downward-sloping or "tilting" response at high frequencies, due to the effects of limited dispersion at high frequencies and greater acoustic absorption.

A completely flat in-room response is therefore usually not desirable and will tend to sound thin or bright. Start with a target curve that follows the natural behavior of your speakers in your room, and then experiment with greater or lesser degrees of tilt in the treble region to obtain the most natural timbral balance.

Low-frequency adjustment

A completely flat response at low frequencies, with complete elimination of peaks due to room modes, may sound light in the bass. Often, a slight increase in the target curve below 100 Hz will give a more balanced sound, yet without introducing audible irregularities in bass response.

Magnitude response dips

In some cases, it may be helpful to adjust the target curve to follow dips in the magnitude response. This can occur where, for example, the listening area is very close to the speakers and the measurements exhibit a dip caused by the vertical response of the speakers themselves. In such a case, adjusting the magnitude response to follow the dip will avoid making the speakers sound worse elsewhere in the room. (You may also wish to try a different set of measurement locations.)

Unlinking left and right channels

Usually, the corresponding left and right channels (front left and right, surround left and right, and rear left and right) should remain linked for target curve adjustment, to ensure that both sides produce the same response across the listening area. In certain unusual circumstances, such as where the magnitude response dip discussed above shows up on only one side, you can try unlinking channels and making separate adjustments.

4.2.4 Saving and loading target curves

To allow you to experiment with different target curves, you can save a target curve to a file and reload it at a later time. Click on the **Save Target** button to save the target curve of the currently displayed channel or group of channels.

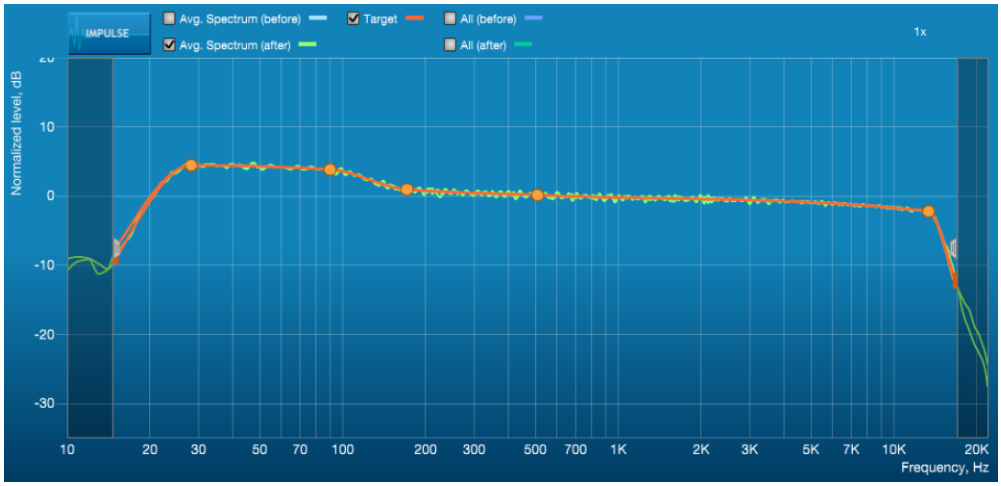
The default location for target files is **C:\Users\[User]\AppData\Roaming\Dirac\OEM\MiniDSP\nanoAVR\Targets**, but you can store them in any location in your file system.

To load a target curve, click on **Load Target**. The currently displayed channel or group of channels will have its target curve updated. (Loading a target will erase the current target curve, so be sure to save it first if needed.)

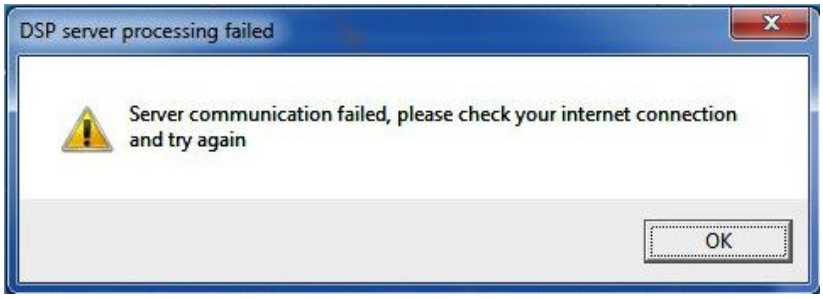
4.3 GENERATING CORRECTION FILTERS

Once you have a target curve set to your satisfaction, click on the **Optimize** button.

The status bar will update as the algorithm progresses. The entire algorithm may take some time to complete, depending on the speed of your computer. When the algorithm completes, the predicted average magnitude response will be shown in green. (The predicted impulse response can be viewed by clicking on the **Impulse** button.)



Dirac Live Calibration Tool For miniDSP will contact the Dirac license server to verify its license, so you will need to be connected to the Internet to perform this step. If a firewall is in place, it must allow HTTP (normal web traffic) to pass. Otherwise, an error such as the following may appear:

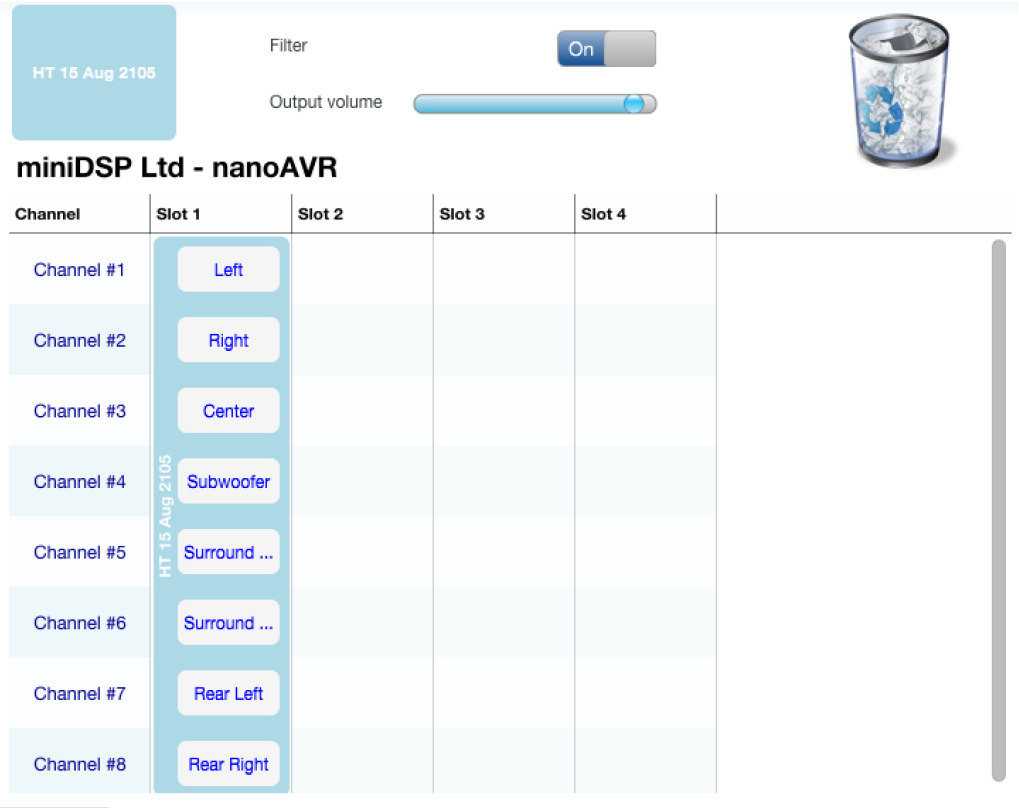


Once the filters are generated, click the **Proceed** button.

4.4 DOWNLOADING AND MANAGING FILTER SETS

The **Export** tab initially shows four empty "slots" for filter sets (a filter set is one filter for every channel). Filter sets are managed with a "drag and drop" metaphor:

- To load the most recently generated filter set into the processor, drag the box at the top left (labeled "HT 15 Aug 2105" in the example) and drop it onto an empty slot.
- To remove a filter set, click on its name (oriented vertically), drag it from the slot and drop it on the trashcan icon at the top right.
- To load a filter set into a slot that already has filters loaded, first delete the loaded filter set by dragging it onto the trashcan icon. Then drag and drop the current filter set onto the now-empty slot.



The two main controls on this tab are:

Filter

Turn this on to enable the Dirac Live® correction filters.

Output volume

Move the slider to adjust the master volume of the processor. Once the computer is disconnected, output volume can be adjusted by the front panel control knob or by an infrared remote control.

5 USING THE DDRC-88A AUDIO PROCESSOR

Once the desired correction filters have been downloaded into the *DDRC-88A* audio processor, the computer is not required and can be disconnected.

5.1 CONFIGURING SOURCE EQUIPMENT

Because of the effect of the Dirac Live algorithm, the settings required in source equipment may differ to the settings used prior to deploying the *DDRC-88A*.

5.1.1 Level trims

Dirac Live aligns the gain of all channels. The level trims of all channels should therefore all be set to the same value (typically 0 dB) in the source equipment.

If you have used the **5.1 Speaker System** or **7.1 Speaker System** configuration and find that your subwoofer is much too loud, it may be that your source equipment has applied a 10 dB LFE alignment gain. A quick solution is to reduce the subwoofer trim level by 10 dB. For more information, see LFE alignment gain on page 44.

5.1.2 Subwoofer channel gain

If your source equipment has a switchable 10 dB gain on the subwoofer output, your choice will interact with the system configuration chosen in the **Output & Levels** tab.

- If you chose the **Custom System** configuration, enable the 10 dB of gain.
- If you chose the **5.1 Speaker System** or **7.1 Speaker System** configuration, disable the 10 dB of gain.

For more information, see LFE alignment gain on page 44.

5.1.3 Bass management

If you are using bass management, follow the instructions of the equipment manufacturer to reconfigure bass management after performing your Dirac Live calibration. The bass management settings that you had prior to including the *DDRC-88A* in your system may no longer be valid due to the changes in system response introduced by Dirac Live.

5.1.4 Delays/speaker distance

Dirac Live calculates the relative delays between all channels and corrects for any differences. In the source equipment, all delays (or equivalently, speaker distance) should therefore be set to the same value , so that the delays calculated by Dirac Live are effective.

5.1.5 Room correction and EQ

Turn off any room correction in the source equipment. Dirac Live performs a full set of sophisticated room correction optimization algorithms, and "adding" additional room correction in the source equipment will not improve the result, and quite possibly make it worse.

EQ that is being used for room correction should also be turned off. If EQ is being used for subjective tuning, note that target curves (see Designing your target curve) can be used to create different filter sets for different

subjective effect. We therefore recommend that target curves be used to tailor the sound on all channels to your liking first, and only then apply EQ in the source equipment if there is a need for rapid "on the fly" subjective EQ adjustments.

5.1.6 Other processing

Any other processing can be applied in the same manner as you did prior to deploying the *DDRC-88A* in your system. In a home theater context, this includes down-mixing (for example from 7.1 to 5.1), up-mixing (including decoders such as Dolby Pro Logic), and "effects" such as hall or ambience simulators. In a studio or sound reinforcement context, effects such as reverberation can continue to be used as before. (Compressor/limiters used to protect power amplifiers should, however, remain connected between the *DDRC-88A* and the power amps.)

5.2 FRONT PANEL

The front panel and/or an infrared remote can be used to control:

- Filter set selection
- Master volume
- Master mute (remote control only)
- Dirac Live® filtering enable/bypass (remote control only)

5.2.1 Status indicators

The current status of the *DDRC-88A* is indicated by a set of LEDs:

Dirac Live Dirac Live® filtering is enabled.

Filter Set Indicates the currently selected filter set (1 through 4).

5.2.2 Front panel controls

The *DDRC-88A* audio processor uses a minimalist physical control design with a single control knob.

To change the volume

Rotate the control knob clockwise to increase the volume, and counter-clockwise to decrease it.

To change the selected filter set

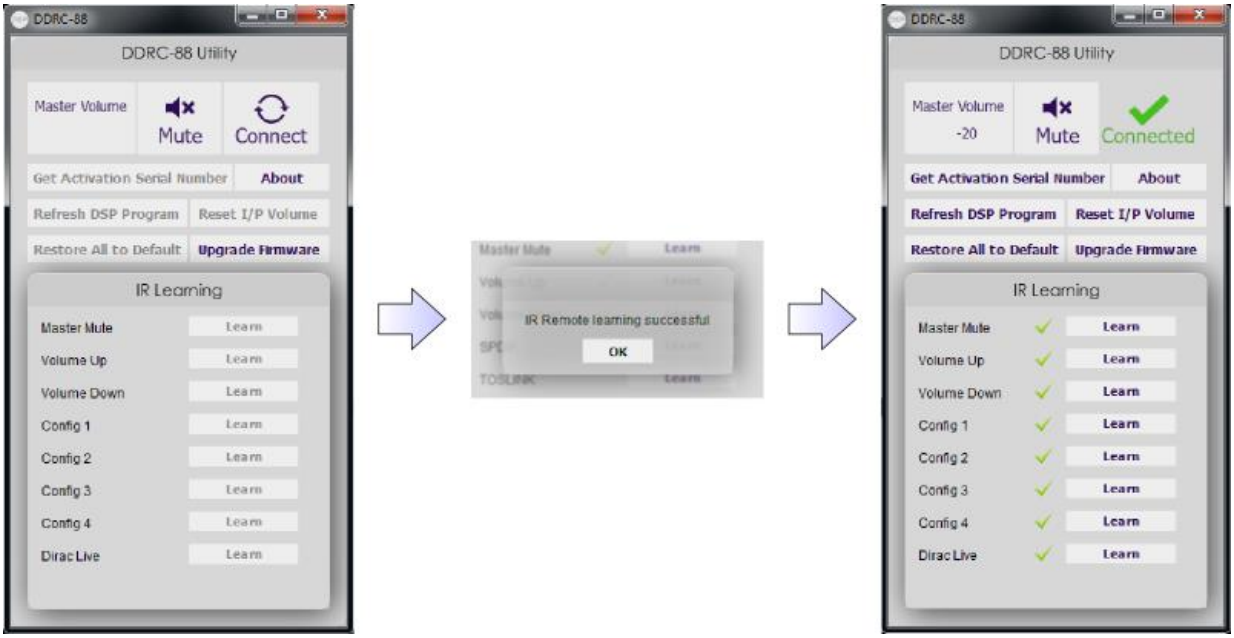
Briefly press the control knob. The current **Filter Set** LED blinks quickly. Rotate the control knob until the desired **Filter Set** LED is blinking. Press the control knob again, and the selected LED will remain steady.

5.3 INFRARED REMOTE CONTROL

Many standard and programmable remote control units can be used with the *DDRC-88A* processor. Instead of adding another remote to your collection, the processor can "learn" the control codes of your current infrared (IR) remote if it supports one of the following remote control codes:

- NEC
- Sony
- Philips RC6
- Apple Remote

Learning is done with the **DDRC-88 Utility** program. After starting the program, click on the **Connect** button. To initiate learning, simply click on one of the function buttons in the **IR Learning** box (**Master Mute**, **Volume Up**, and so on). Then point your remote at the processor and click on the button that you want to use for that function. A dialog will appear to show that the code was recognized. (If the processor does not recognize the remote control code, then it will time out and display a message saying that IR learning failed.)



Once programmed, check that the programmed buttons perform the expected function.

6 GAIN STRUCTURE

When deploying a DSP solution in your audio system, a topic that becomes more important than with analog equipment is *gain structure*. This means that the signal levels throughout the system should be set high enough to maximize digital resolution and minimize noise, but not so high as to result in clipping and distortion.

The *DDRC-88A* provides a great deal of flexibility in matching with other equipment. In addition to providing balanced and unbalanced inputs and outputs, input sensitivity and output gain can be selected to adapt to different equipment. The *DDRC-88A* ships with a conservative gain structure that will rarely result in clipping, but you may wish to adjust these settings for your own installation. This section explains gain structure and how to optimize it.

6.1 GAIN STRUCTURE OVERVIEW

Figure 6.1 below illustrates the elements that affect gain and gain structure (for a single channel). The source/processor has a master volume control and typically a user-settable trim control on each output channel. It may also have an additional 10 dB gain on the LFE channel. The *DDRC-88A* has a master volume control and an internal trim control (not visible to the user). In addition, there are per-channel settings for input sensitivity and output gain. Finally, the power amplifier itself may have a master gain control or a level trim on each channel.

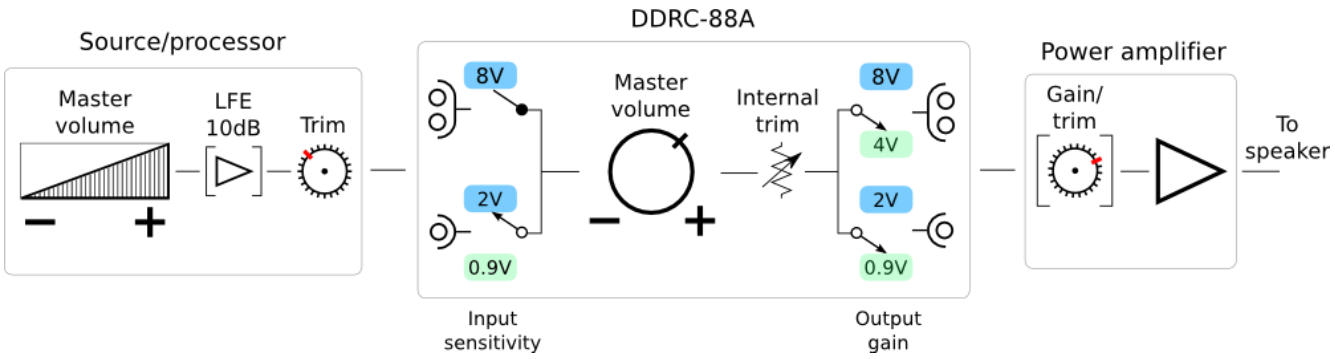


Figure 6.1. Overview of gain structure through the system

6.2 CHOOSING GAIN STRUCTURE SETTINGS

The *input sensitivity* is the voltage that generates a full-scale digital input signal to the *DDRC-88A*'s DSP (digital signal processor). The *output gain* setting specifies the voltage that results from a full-scale digital output signal from the DSP.

The best choice of input sensitivity and output gain settings can be determined in two ways: by the output level of the source and the input sensitivity of the amplifier, or by the resulting input-output gain of the *DDRC-88A*. We suggest initially using Figure 6.2 as follows:

1. Knowing the maximum output level of your source equipment, run a line across to either *Unbalanced input* or *Balanced* input. The best input sensitivity is the one that your line intersects. If you are in the red zone, limit the master volume control on your source equipment to avoid overloading the *DDRC-88A* input.
2. Compare the maximum output level of your source equipment with the input sensitivity of your amplification, and choose the lowest number. Then run a line across to either *Unbalanced output* or *Balanced output*. The best output gain setting is the one that your line intersects. (With this recommendation we allow the *possibility* that the *DDRC-88A* may overdrive the amplifier. Since Dirac Live will tend to reduce output levels somewhat, over-driving it is unlikely in practice.)

If your equipment offers a choice of unbalanced or balanced inputs or outputs, this diagram can also be used to help choose which type of connection to use. Note that there is no sensitivity adjustment for balanced inputs (the only available choice is 8 VRMS). In addition, if an input channel is set for 0.9V sensitivity, a balanced connection can **not** be made to that channel.

All input channels must have the same type of connection (balanced or unbalanced) and the same input sensitivity setting. (Output channels can have a mix of balanced and unbalanced connections and a mix of output gain settings. If using a mix, always do a Gain structure optimization.)

Changing connection type or gain structure settings of individual channels will invalidate all of your existing Dirac Live projects. Therefore, if making a change to gain structure, you must either change all channels in the exact same way, or completely redo your Dirac Live calibration afterwards.

If, after performing an initial calibration with **Dirac Live Calibration Tool for miniDSP**, you find that the input-output gain is not suitable for your system, then refer to Figure 6.3 to choose different settings that match your desired input-output gain.

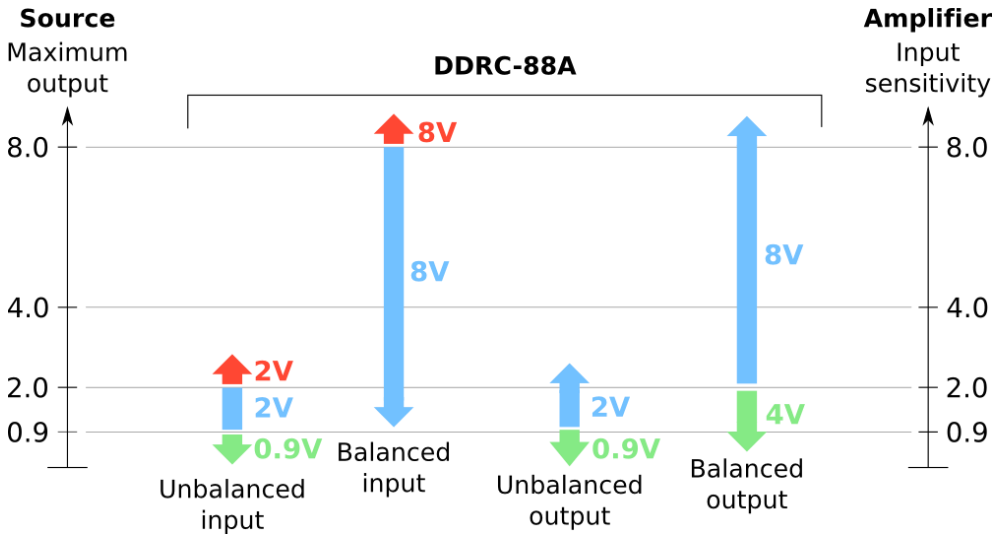


Figure 6.2. Gain structure settings based on input and output levels

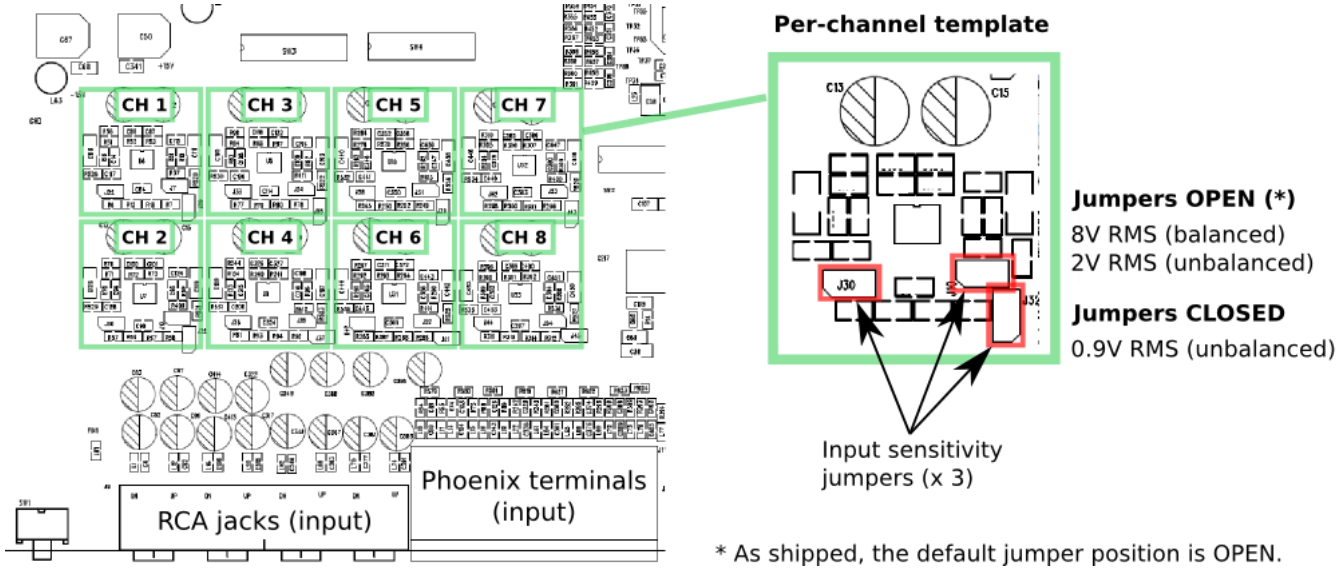
		Unbalanced output		Balanced output	
	Setting	0.9V	2V	4V	8V
Unbalanced input	0.9V	0 dB	6 dB	12 dB	18 dB
	2V	–6 dB	0 dB	6 dB	12 dB
Balanced input	8V	–18 dB	–12 dB	–6 dB	0 dB

Gain values are simplified to give exact multiples of 6dB.

Figure 6.3. Input-output gain determined by connection type and gain structure settings

6.2.1 To change input sensitivity

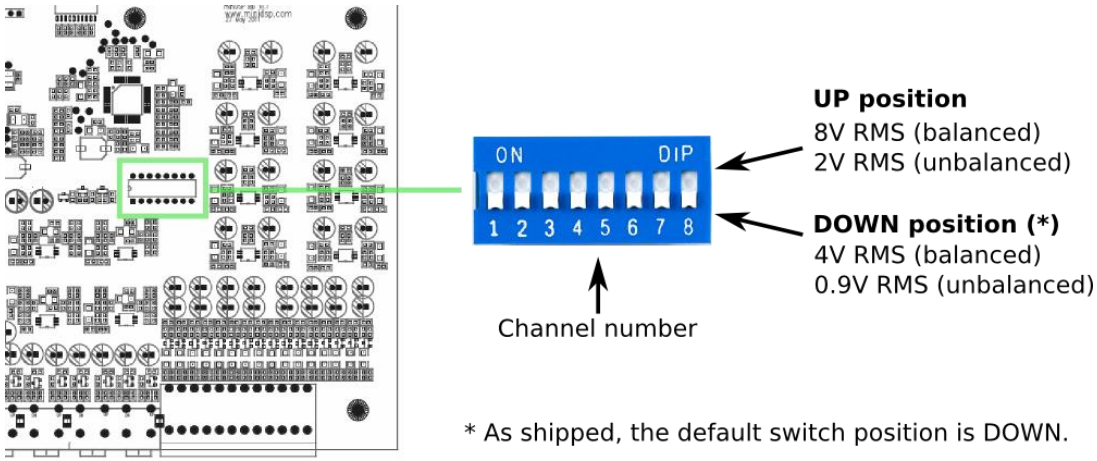
Input sensitivity can be changed for **unbalanced connection only**, by moving a set of three jumpers on each channel. The diagram below illustrates the location of the jumper headers for each channel. For each channel that is to be changed, lift the jumpers and replace them in either the OPEN position (jumper pushed onto one pin only) or CLOSED position (jumper pushed onto both pins).



Do not make a balanced connection to the input of any channel with jumpers in the CLOSED position.

6.2.2 To change output gain

A set of eight DIP switches located on the main circuit board select the output gain of each channel. The switches can be set independently for each channel, and affect output gain for both the balanced and unbalanced output connections to that channel.



6.3 Optimizing gain structure

If the acoustic output level of different channels shows substantial differences (greater than 6 dB), then there may be room to improve the gain structure of your system. Follow the procedure below carefully. Note that:

- You must not adjust the **Channel volume** sliders away from the maximum at any point in this procedure.
- It is not necessary to get all test meter readings at exactly the target level. The aim of the procedure is simply to reduce gain differences between channels, not to eliminate them.
- In step 6, there is a judgment call on which channels should have increased gain and which should have reduced gain.
- It is possible that you will get to step 7 and realize that there is nothing you can do to improve the gain structure of your system.

Changing the connection type or gain structure settings of individual channels will invalidate all of your existing Dirac Live projects. You will need to discard all existing projects and filter sets after completing this procedure, and redo your Dirac Live calibration.

6.3.1 Procedure for optimizing output gain

1. If your amplifiers have gain controls, set them all to maximum (including the subwoofer).
2. Quit and restart **Dirac Live Calibration Tool**.
3. On the **Output & Levels** tab, set all **Channel volume** sliders to maximum. Do not move them again while performing this procedure.
4. Set the **Output volume** low and turn on the test signal for channel 1. Adjust **Output volume** and **Input gain** so the meter reads in the middle of the green area i.e. −12 dB.
5. Turn on the test signal for the remaining channels and note the level of the test signal on the meter (for example, −6 dB, −15 dB, etc).
6. Determine the target level for each channel:
   a. The speaker channels should all have the same target level.
   b. If the **Custom System** configuration is used, the subwoofer target is the same as the speakers.
   c. If the **5.1** or **7.1** configuration is used, the subwoofer target is 6 dB higher than the speakers. For example, if the speaker target is −12 dB, the subwoofer target is then −6 dB.
7. Using the tables on the next page, decide on the best method to change the gains of each channel to get closer to the target values.
8. Power off the power amplifiers and the *DDRC-88A*, and make the changes.
9. Power on the *DDRC-88A* and the power amplifiers, restart **Dirac Live Calibration Tool**, and re-run the levels check on the **Output & Levels** tab.

6.3.2 To increase output gain

Before	After	Gain change (approx.)
DIP switch DOWN	Dip switch UP	+6 dB
Unbalanced connection	Balanced connection (*4)	+12 dB (*5)
Unbalanced connection	Balanced to unbalanced adapter (*6)	+6 dB

*4. This change can be made only if your amplifier has balanced inputs.

*5. This gain change applies in the typical case in which the amplifier has the same sensitivity/gain on both balanced and unbalanced input connections. Some amplifiers have half the sensitivity on the balanced inputs, in which case this connection change will result in a +6 dB gain change. Please check the specifications of your amplifier.

*6. See page 48 for wiring.

6.3.3 To reduce output gain

Before	After	Gain change (approx.)
Amplifier gain full	Amplifier gain reduced	–variable
DIP switch UP	Dip switch DOWN	–6 dB
Balanced connection	Unbalanced connection (*7)	–12 dB (*8)
Balanced connection	Balanced to unbalanced adapter (*7,*9)	–6 dB (*10)

*7. This change can be made only if your amplifier has unbalanced inputs.

*8. This gain change applies in the typical case in which the amplifier has the same sensitivity/gain on both balanced and unbalanced input connections. Some amplifiers have half the sensitivity on the balanced inputs, in which case this connection change will result in –6 dB gain change. Please check the specifications of your amplifier.

*9. See page 48 for wiring.

*10. This gain change applies in the typical case in which the amplifier has the same sensitivity/gain on both balanced and unbalanced input connections. Some amplifiers have half the sensitivity on the balanced inputs, in which case this connection change will have no useful effect. Please check the specifications of your amplifier.

6.4 LFE ALIGNMENT GAIN

> "Known as the LFE channel (for Low Frequency Effects), it handles bass created specifically for subwoofer boom effects and may also carry low frequency information from the other channels in order to enrich the overall soundtrack."
>
> *5.1-Channel Production Guidelines*, Dolby Laboratories Inc.

> "The LFE channel is recorded with a level offset of −10 dB. This offset is compensated for in the reproduction system, where the LFE loudspeaker has an acoustic output (within its low frequency passband) of +10 dB with respect to the other channels."
>
> *Recommendation ITU-R BR.1384-2 : Parameters for international exchange of multi-channel sound recordings with or without accompanying picture*, International Telecommunication Union.

6.4.1 The LFE channel and the DDRC-88A

As noted above, the LFE channel is recorded 10 dB lower than the speaker channels. On replay, a signal in the LFE track must therefore produce 10 dB more *acoustic* output than the same signal in one of the speaker tracks. If we were to assume that the monitoring system (amplifiers, speakers, and subwoofer) were initially calibrated so that the same electrical input signal delivered the same acoustic output from all channels, then the channel that plays the LFE content (i.e. the subwoofer channel) must have 10 dB of electrical gain added to it, somewhere between the bitstream decoder and the subwoofer driver.

This additional 10 dB gain we will refer to as the "LFE alignment gain," since it is there to align the level of LFE acoustic output with the output from the speakers. (Bass management is discussed below.)

In a studio environment, where calibration of the monitoring system is likely, the mixing desk typically includes an additional 10 dB of gain between the recording output and the monitoring output. With that said, subwoofers designed for studio use often include an "LFE +10 dB" gain switch, specifically to cater for those situations in which the mixing desk does not have this facility.

In a home theater environment, *a priori* calibration of the amplifier/speaker system is unlikely. Nonetheless, A/V receivers and processors typically apply 10 dB of gain to the subwoofer output. In other cases, such as players with analog output, the source does not apply the LFE alignment gain and the subsequent electronics (receiver or subwoofer amplifier) is expected to provide it instead. SPL calibration is done for the whole replay chain, and may be a manual process involving test signals and adjusting subwoofer gain, or an automated process whereby onboard software adjusts the subwoofer trim level.

The *DDRC-88A* caters to all of these situations. If the *DDRC-88A* is to "ignore" the whole question of LFE alignment gain, choose the **Custom System** configuration. Every channel will be treated as equal (in terms of gain) by Dirac Live, and the onus is on external equipment to provide 10 dB of alignment gain – usually in the source equipment, but it can also be done in the subwoofer amplifier after Dirac Live calibration has been completed.

If, on the other hand, the *DDRC-88A* is to be "aware" of the need for LFE alignment gain, then choose the **5.1 Speaker System** or **7.1 Speaker System** configuration. In this case, Dirac Live will calibrate for 10 dB higher acoustic output on the subwoofer channel by adjusting its internal trim on the subwoofer channel. This does *not* necessarily mean that the *DDRC-88A* will apply 10 dB of gain to its subwoofer channel – that is determined by the calibration process.

6.4.2 LFE effect on gain structure

In the gain structure optimization procedure on page 42, we recommend that for the **5.1 Speaker System** or **7.1 Speaker System** configurations, the target level of the subwoofer be set 6 dB higher (*1) than the speaker channels. This will push the bulk of the 10 dB LFE alignment gain into the subwoofer amplifier, which will in turn result in an input-output gain on the *DDRC-88A* subwoofer channel of about unity. (Assuming the input sensitivity is the same as the output gain setting.)

Purely in terms of gain structure, this is better than applying the extra gain in the source equipment. This is illustrated in Figure 6.4 below. At the top, with 10 dB of gain applied in the source/processor, the subwoofer signal is 10 dB higher than the speaker signal at the *DDRC-88A* inputs and outputs. At the bottom, with the 10 dB extra gain applied in the subwoofer amplifier, we still have the required 10 dB greater subwoofer output, but the subwoofer signal at the *DDRC-88A* inputs and outputs is aligned with the level on the speaker channels.

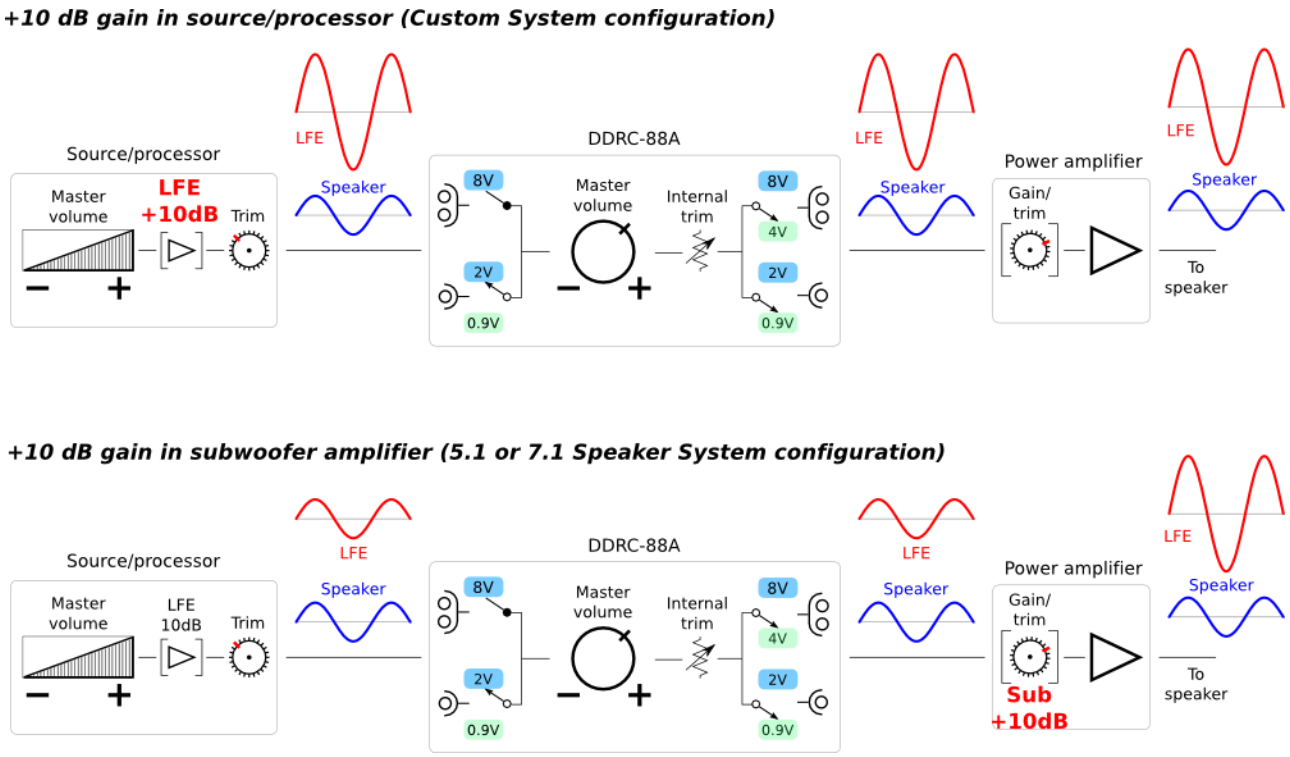


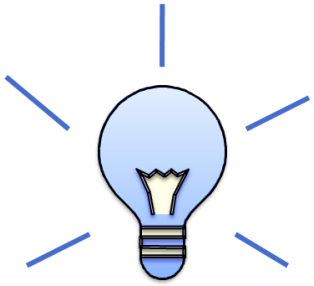
Figure 6.4. Illustrating signal levels depending on where LFE alignment gain is applied

(*1. The suggested 6 dB higher target level is only an approximation—the resulting input-output gain of the subwoofer channel depends on the bandwidth of the subwoofer and its measured in-room response. A more precise setting would require measuring the electrical output of the *DDRC-88A* after calibration, but this is unlikely to be necessary nor it is something that can be covered in this manual.)

6.4.3 LFE gain structure problems

The difference in signal levels noted above for **Custom Configuration** is likely not a problem in practice, in most systems. If, however, you do experience issues with distortion on the subwoofer channel at high playback levels, it may be that you are clipping the subwoofer output of the source equipment or the subwoofer input of the *DDRC-88A*. Here are two possible remedies:

- Reduce the level of the subwoofer trim in the source equipment and increase the gain on the subwoofer amplifier by the same amount. If you do this, you will need to remember to set the subwoofer gain back to its previous position if you redo your Dirac Live calibration.
- Remove the 10 dB alignment gain from the source equipment completely, and use the **5.1 Speaker System** or **7.1 Speaker System** configuration. If your source does not have a 10 dB gain switch/setting, you should be able to accomplish the same thing by setting the subwoofer trim control to −10 dB. You will need to redo the gain structure optimization procedure on page 42 and your Dirac Live calibration as described in sections 3 and 4.



6.4.4 Bass management and LFE alignment gain

When redirected bass from the speaker channels is mixed with the LFE channel, it is mixed at a relative level of −10 dB. After 10 dB LFE alignment gain is applied to the mixed signal, the redirected bass is at the correct level again. The whole discussion above therefore applies regardless of whether the subwoofer output from the source equipment is carrying only the LFE content, or bass-managed content that includes the LFE content mixed with bass from the speaker channels.

If mixing of redirected bass is performed digitally, there is a danger of clipping the signal. The LFE signal already uses the full-scale digital range, so adding in bass from another seven channels, even reduced by 10 dB, could exceed the maximum digital signal level. To avoid this, the LFE is reduced prior to mixing by 5 dB and the redirected bass from the speaker channels is reduced by 15 dB. After D/A conversion, 15 dB of gain is applied in the analog domain (10 dB alignment gain, plus another 5).

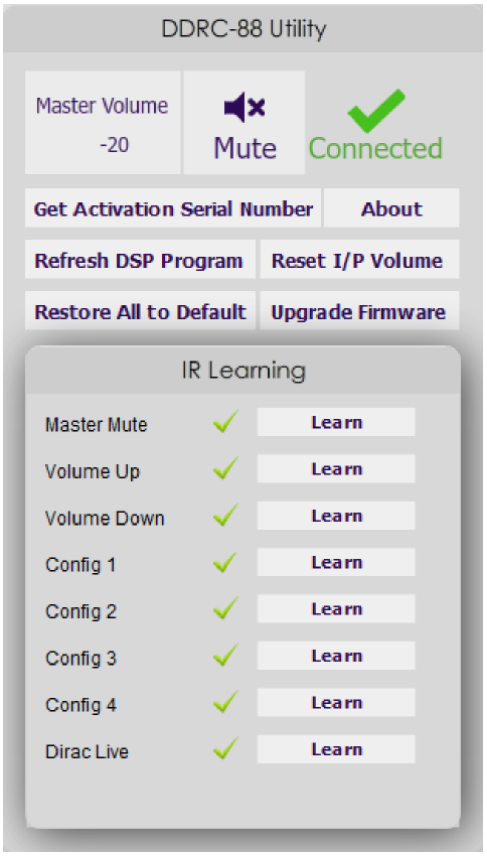
Some consumer equipment, however, does not apply the 15 dB of analog gain. This is more likely in front-end equipment like DVD/Blu-ray players. If a unit like this is connected directly to the *DDRC-88A* inputs and bass management is used, you will need to:

- Use the **5.1 Speaker System** or **7.1 Speaker System** configuration for 10 dB of LFE alignment gain, and
- Add another 5 dB of gain by:
    - Raising the subwoofer trim level in the source/processor by 5 dB, or
    - Adding 5 dB of gain in the subwoofer amplifier after completing Dirac Live calibration.

7 Additional Information

7.1 The DDRC-88 Utility program

The DDRC-88 Utility program is an auxiliary program for configuring certain functions of the *DDRC-88A*. It is not used for measurement or calibration.



Master volume

Displays the current master volume setting.

Mute

Mutes all audio output.

Connect/Connected

Click on the **Connect** icon to connect to the *DDRC-88A*. To disconnect, quit from the **DDRC-88 Utility**.

Get Activation Serial Number

Click to get the unique activation serial number for this unit. See License activation on page 12.

Refresh DSP Program

Updates the internal DSP program with the version contained in the **DDRC-88 Utility**. Use this after downloading a new version of **DDRC-88 Utility**.

Restore All to Default

Clears all four loaded filter presets.

About

Provides the current versions of DSP firmware. Please provide this information when submitting a support ticket at minidsp.desk.com.

Reset I/P Volume

Resets all **Channel volume** sliders to the default.

Upgrade Firmware

Starts the MCU firmware upgrade process. See MCU firmware update on page 51.

IR Learning

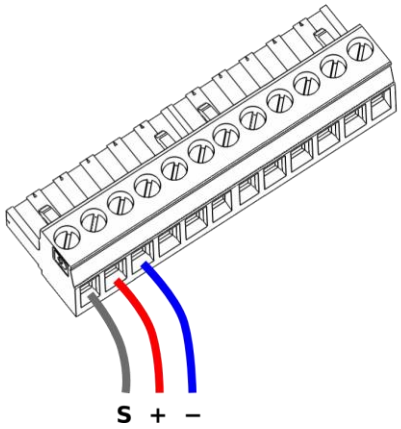
Buttons used to learn infrared remote control codes. See Infrared remote control on page 37.

Be sure to quit **Dirac Live Calibration Tool For miniDSP** before starting the **DDRC-88 Utility** program. Running both programs at the same time may result in communication conflicts and odd behavior.

7.2 BALANCED WIRING TIPS

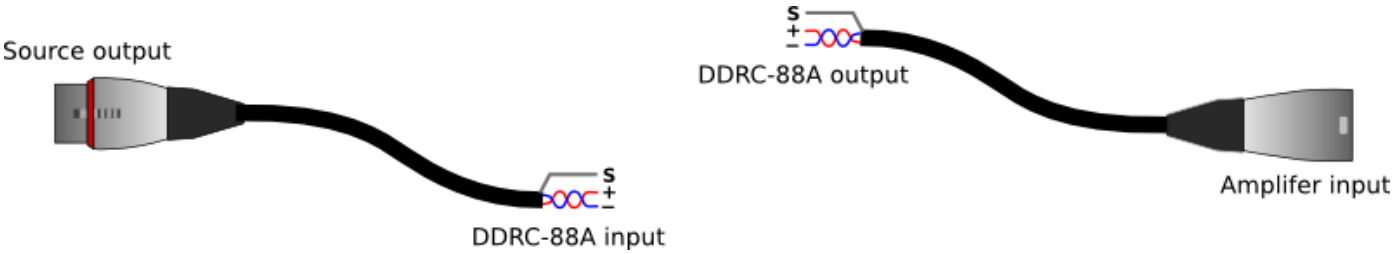
7.2.1 Phoenix terminal blocks

To make balanced connections, remove the terminal blocks from the rear of the *DDRC-88A*. Connect individual wires from a shielded pair cable to each set of screw terminals as shown below. After all connections to the terminal block are secure, firmly re-insert the terminal blocks.



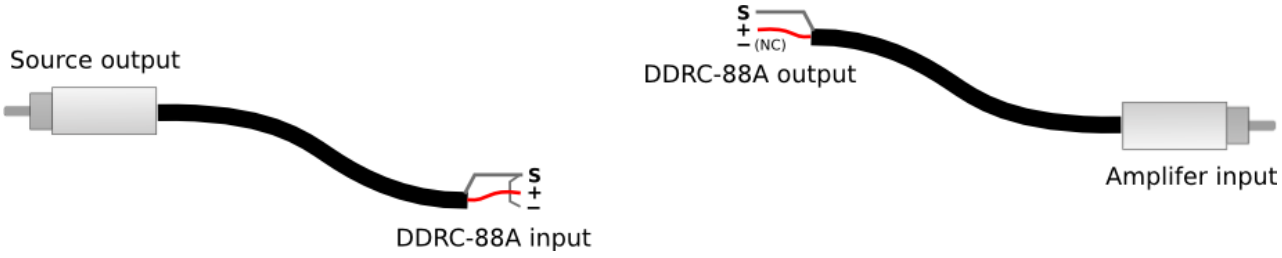
7.2.2 XLR adapters

If connecting the *DDRC-88A* to equipment that uses XLR connectors, suitable cabling can be made by cutting XLR-XLR or microphone cables in two. The two halves are then used for input and output, as shown here:



7.2.3 RCA adapters

In certain special cases, it can be helpful to connect a balanced input or output of the *DDRC-88A* to an unbalanced (RCA) connection on the source or power amplifier. The connection scheme is shown below. Note that the cable shield must be connected to the '–' terminal on the *DDRC-88A* input, whereas on the output side, the '–' terminal is not connected.



7.3 SPECIFICATIONS

Computer connectivity	Driverless USB 2.0 control interface for Windows and Mac OS X
Analog inputs	8 x Balanced (Terminal Block) or 8 x Unbalanced (RCA) inputs • Maximum balanced input voltage: 8 VRMS • Maximum unbalanced input voltage: 2.0 or 0.9 VRMS (jumper selectable)
Analog outputs	8 x Balanced (Terminal Block) and 8 Unbalanced (RCA) outputs • Maximum balanced output voltage: 8.0 or 4.0 VRMS (DIP switch selectable) • Maximum unbalanced output voltage: 2.0 or 0.9 VRMS (DIP switch selectable)
Audio sample rate / Resolution	Input/output resolution: 24-bit integer Sample rate: 48 kHz
Audio processing	32-bit floating-point processor
Storage/Presets	4 filter sets stored, selectable from front panel or IR remote
Infrared remote	"Learning remote" capabilities (NEC, Philips, Sony, Apple) Controls master volume, mute, filter set selection, Dirac Live® filtering enable
Power supply	12 VDC single supply
Dimensions (H x W x D)	41.5 x 429 x 252 mm; 1RU height

7.4 TROUBLESHOOTING

The following table lists the most common causes of issues with the *DDRC-88A*. If following this table does not provide a solution, see Obtaining Support below.

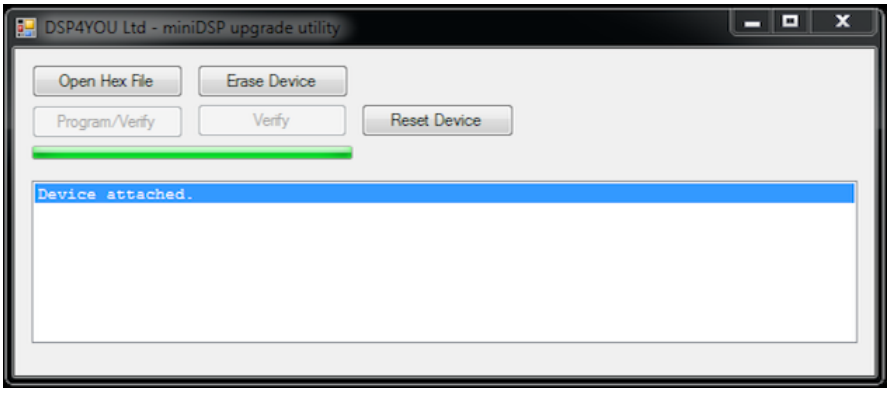
Item#	Symptoms	Troubleshooting recommendation
1	Cannot install software	a. Confirm that you downloaded and installed the required frameworks first (see Software Installation).
2	The **DDRC-88 Utility** cannot connect to the *DDRC-88A* hardware unit	a. Check that the USB cable to the *DDRC-88A* is firmly connected b. Check that you are not running **Dirac Live Calibration Tool For miniDSP** (the two programs cannot be used at the same time).
2	The Dirac activation screen does not recognize the serial number	a. Do not use the serial number printed on the label of the unit. You must use the serial number obtained from the firmware using the **DDRC-88 Utility** program.
3	The license validation screen doesn't accept my username and password	a. The "username" must be the **email address** that you used when activating your license on the Dirac Live activation screen. Check that you are using the same email address and password.
4	The *DDRC-88A* doesn't appear in the **Sound System** tab	a. Check that the USB cable to the *DDRC-88A* is firmly connected. b. Check that you do not have any other program running that is attempting to communicate with the *DDRC-88A*, such as the **DDRC-88 Utility** program. c. Check that you have the miniDSP version of the software installed, called **Dirac Live Calibration Tool For miniDSP**. d. Go to the **Sound System** tab and click the **Rescan** button.
5	The measurement test signal produces no output	a. Ensure that the *DDRC-88A* processor is connected correctly into the audio system. b. Check that the downstream amplification is powered on. c. Check that the downstream amplification is not muted and doesn't have gain/trim controls set to zero.
6	No input from measurement microphone	a. Check that the USB cable to the UMIK-1 is securely seated. b. Check that the UMIK-1 is selected in the **Mic Config** tab.
7	Insufficient recording level	a. Increase microphone level in the **Output & Levels** tab. b. Go to the Control Panel and view the Recording tab of the Sound pane. Select the UMIK-1 and view its Properties. In Levels, set the gain to 100. c. Increase system output volume.

8	Unable to generate correction filters (**Optimize** button)	a. Check that your computer is connected to the Internet and able to pass HTTP (web) traffic. b. Check that you do not have any other program running that is attempting to communicate with the *DDRC-88A*, such as the **DDRC-88 Utility** program.
9	No audio playthrough	a. Check that the *DDRC-88A* processor is not muted. b. Check that the *DDRC-88A* processor master volume control is not turned down. c. Check that all cables are securely seated. d. Check that cables are connected to the correct inputs and outputs on the rear panel. e. Check that the downstream amplification is powered on, is not muted, and doesn't have gain/trim set to zero. f. Check that the source equipment is not muted and does not have volume set to zero.

7.5 MCU FIRMWARE UPDATE

miniDSP may occasionally provide an update to the DDRC-88A MCU firmware to enable new features. To update the MCU firmware:

1. Download and install the latest **DDRC-88 Utility** package from the **User Downloads** section of the miniDSP website.
2. Connect the DDRC-88A to your computer via USB.
3. Start the **DDRC-88 utility** program.
4. Click on the **Connect** button. The button will display a green tick if connection is successful.
5. Click on the **Upgrade Firmware** button. The DDRC-88A will be put into boot loader mode and the miniDSP upgrade utility will start. The status line should display "Device attached".



4. Click on the **Open Hex File** button and select the .hex file included in the download. It will have a name like **ENC_DDRC88_Ver2.17_SS.hex**.
5. Click on the **Program/Verify** button. The status bar will indicate progress. **Do not disconnect** the DDRC-88A during the firmware upgrade!
6. Click on the **Reset Device** button. The processor will return to normal operating mode, and can be used as normal.

DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE *DDRC-88A* WHILE FIRMWARE UPDATE IS IN PROGRESS. DOING SO MAY "BRICK" YOUR *DDRC-88A*.

7.6 OBTAINING SUPPORT

1. Check the forums on miniDSP.com to see if this issue has already been raised and a solution or solutions provided.
2. Contact miniDSP via the support portal at minidsp.desk.com with:
   a. The product information obtained from **DDRC-88 Utility** (**About** button) and **Dirac Live Calibration Tool for miniDSP** (**Sound System** tab).
   b. A clear explanation of the symptoms you are seeing.
   c. A description of the troubleshooting steps you performed (see Troubleshooting above) and the results obtained.